COLLEGE OF ENGINEERING, GUINDY
ANNA UNIVERSITY
CHENNAI 600025

EC5612 - WIRELESS COMMUNICATION AND NETWORKING LABORATORY

SEMESTER VI (R-2019)
(January 2024 - May 2024)

RECORD

SUBMITTED BY:

SARAN A

2021105325

In partial fulfillment for the award of the degree of
BACHELOR OF ENGINEERING
IN
ELECTRONICS AND COMMUNICATION ENGINEERING

ANNA UNIVERSITY
COLLEGE OF ENGINEERING GUINDY

<u>BONAFIDE CERTIFICATE</u>

NAME: SARAN A
DEPT: B.E. Electronics and Communication Engineering
REGISTER NUMBER: 2021105325

It is certified that this is the bonafide record of the work done by the above-mentioned student in EC5612 - WIRELESS COMMUNICATION AND NETWORKING LABORATORY - SEMESTERVI (R-2019) during the period January 2024 - May 2024.

Signature of Lab-In-Charge

Signature of the Head of the Department

Submitted for the practical exam held on _____________
Signature of Internal Examiner

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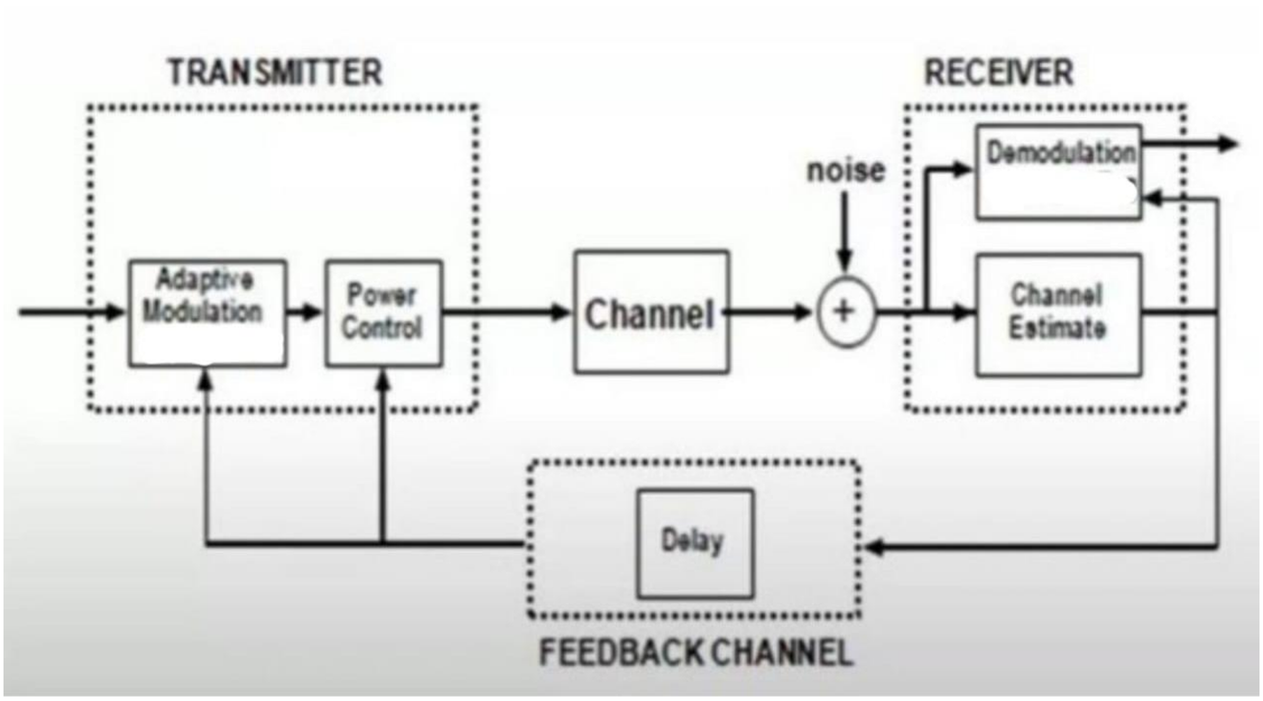
BLOCK DIAGRAM:



EXP NO: 01 **DATE:** 31/01/2024	**Performance studies of Adaptive Modulation**

AIM:

To perform Adaptive modulation and observe its characteristics.

SOFTWARE REQUIRED:

MATLAB R2022b

THEORY:

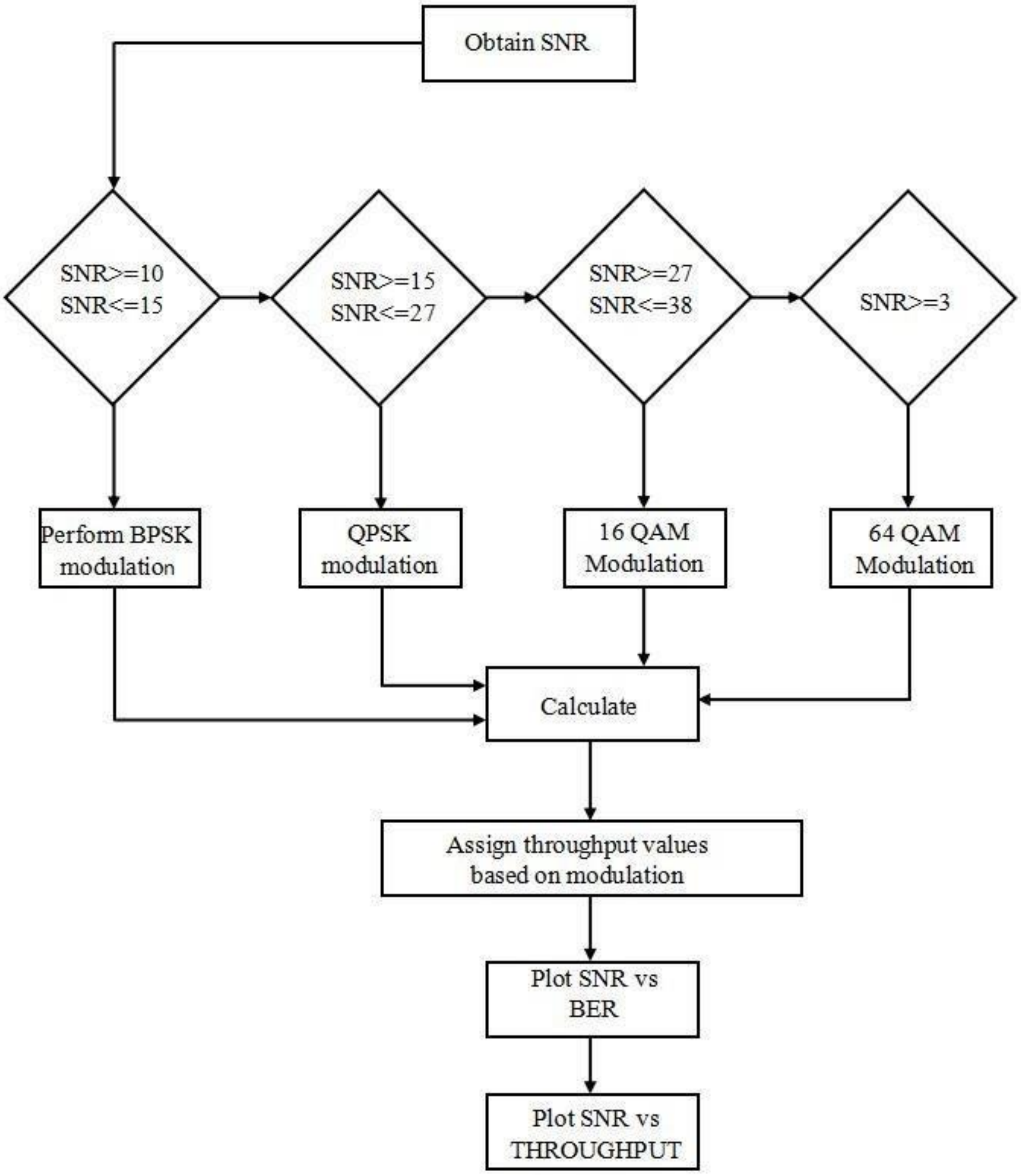
WHAT IS ADAPTIVE MODULATION?

- Adaptive modulation is a technique which allows a radio to change its speed (Modulation rate) as conditions in the radio network changes.
- Interference from outside sources, such as changes in the environment (temperature, tree foliage, moving objects) all effect radio coverage.
- Adaptive modulation enables robust and spectrally efficient transmission over time varying channels.
- The basic premise is to estimate the channel at the receiver and feed this channel back to the transmitter so that the transmission can be adapted relative to channel characteristics.
- Thus, the system can be designed efficiently for the worst-case channel characteristics.

WHY ADAPTIVE MODULATION OVER MODULATION?

- In modulation, there is inefficient utilization of channel and wastage of power which is not the case with adaptive modulation,
- With AMC, data rate and signal power are varied as per the channel conditions for efficient utilization of resources whereas in modulation, they remain fixed.
- AMC guarantees 99% of data transfer even at worst channel condition without any wastage of resources which is not the case with modulation.

FLOWCHART:


Obtain SNR
SNR>=10 SNR<=15
SNR>=15 SNR<=27
SNR>=27 SNR<=38
SNR>=3
Perform BPSK modulation
QPSK modulation
16 QAM Modulation
64 QAM Modulation
Calculate
Assign throughput values based on modulation
Plot SNR vs BER
Plot SNR vs THROUGHPUT

ADVANTAGES AND DISADVANTAGES:

ADVANTAGES:

- AMC allows the system to overcome fading and other interference.
- Maximize the throughput.

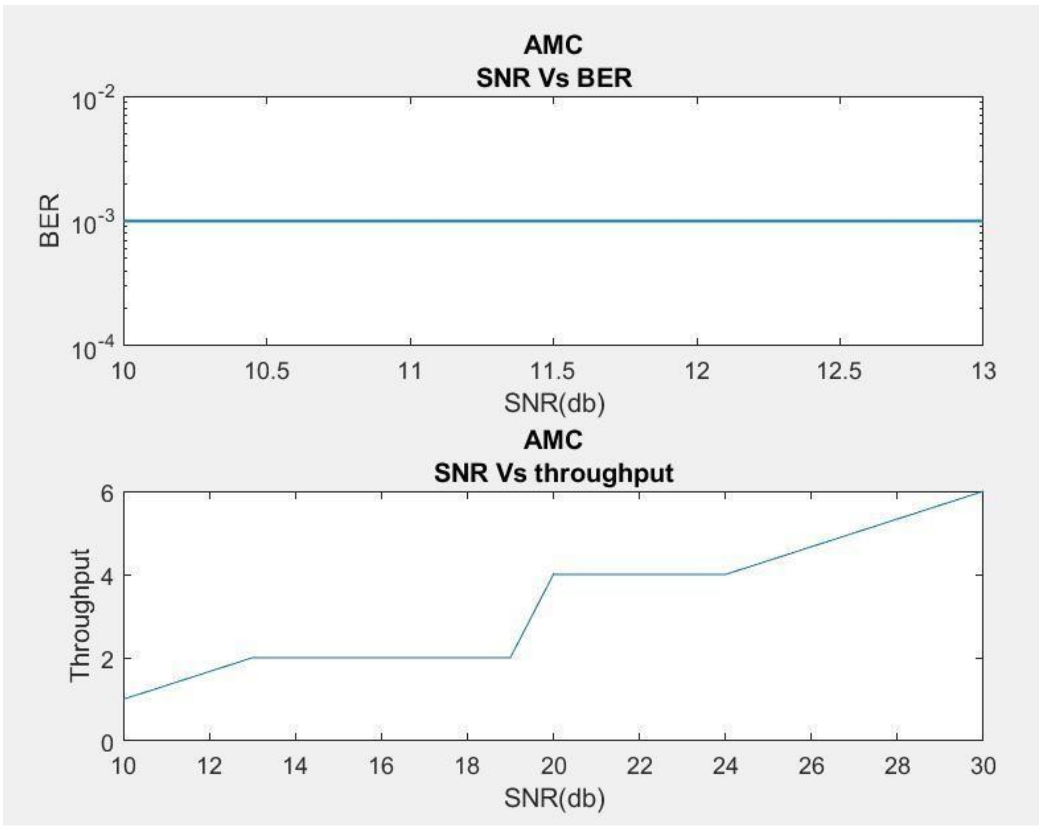
DISADVANTAGES:

- AMC is sensitive to measurement error and delay.
- It requires a feedback system between transmitter and receiver which may not be feasible for some system.
- There are possibilities of wastage of system capacity
- High complexity

ALGORITHM:

1. Define a random sequence of bits for BPSK, QPSK, 16PSK, 64PSK.
2. SNR Range is also defined.
3. If 10<=SNR<=15 use BPSK modulation.
4. If 15<=SNR<=27 use QPSK modulation.
5. If 27<=SNR<=38 use 16PSK modulation.
6. If SNR >38 use 64 PSK Modulation.
7. If BER difference is zero BER=10^-7.
8. Plot a graph with X axis: SNR , Y axis:BER.

MATLAB OUTPUT:



MATLAB CODE:

```
clc;
clear all
close all
snr= randi([10,30],1,15);
snr=sort(snr);
err=[];
for i=1:length(snr)
if(snr(i)>=0 && snr(i)<13)
a=randi([0,1],1,1000);b=pskmod(a,2);
end
if(snr(i)>=13 && snr(i)<20)
a=randi([0,3],1,1000); b=pskmod(a,4);
end
if(snr(i)>=20 && snr(i)<26)
a=randi([0,15],1,1000); b=qammod(a,16);
end
if(snr(i)>=26 && snr(i)<=30)
a=randi([0,63],1,1000);b=qammod(a,64);
end
N0=1/10^(snr(i)/10);
g=awgn(b,snr(i));
n=sqrt(N0/2)*(randn(1,length(a))+1i*randn(1,length(a)));
f=g+n;
if(snr(i)>=10 && snr(i)<13)
d=pskdemod(f,2);
end
if(snr(i)>=13 && snr(i)<20)
d=pskdemod(f,4);
end
if(snr(i)>=20 && snr(i)<26)
d=qamdemod(f,16);
end
if(snr(i)>=26 && snr(i)<=30)
d=qamdemod(f,64);
end
[n,r]=biterr(a,d);

if(r==0)
r=1e-7; end
err=[err r];
end
subplot(2,1,1);semilogy(snr,err,'linewidth',1);
xlabel('SNR(db)');
ylabel('BER'); title({'AMC';'SNR Vs BER'});
thruput=[];
for i=1:length(snr)
if(snr(i)>=10 && snr(i)<13)
thruput=[thruput 1]; end
if(snr(i)>=13 && snr(i)<20)
thruput=[thruput 2]; end
if(snr(i)>=20 && snr(i)<=25)
thruput=[thruput 4]; end
if(snr(i)>=26 && snr(i)<=30)
thruput=[thruput 6];
end
end
subplot(2,1,2); plot(snr ,thruput);
xlabel('SNR(db)'); ylabel('Throughput');
title({'AMC';'SNR Vs throughput'});
```

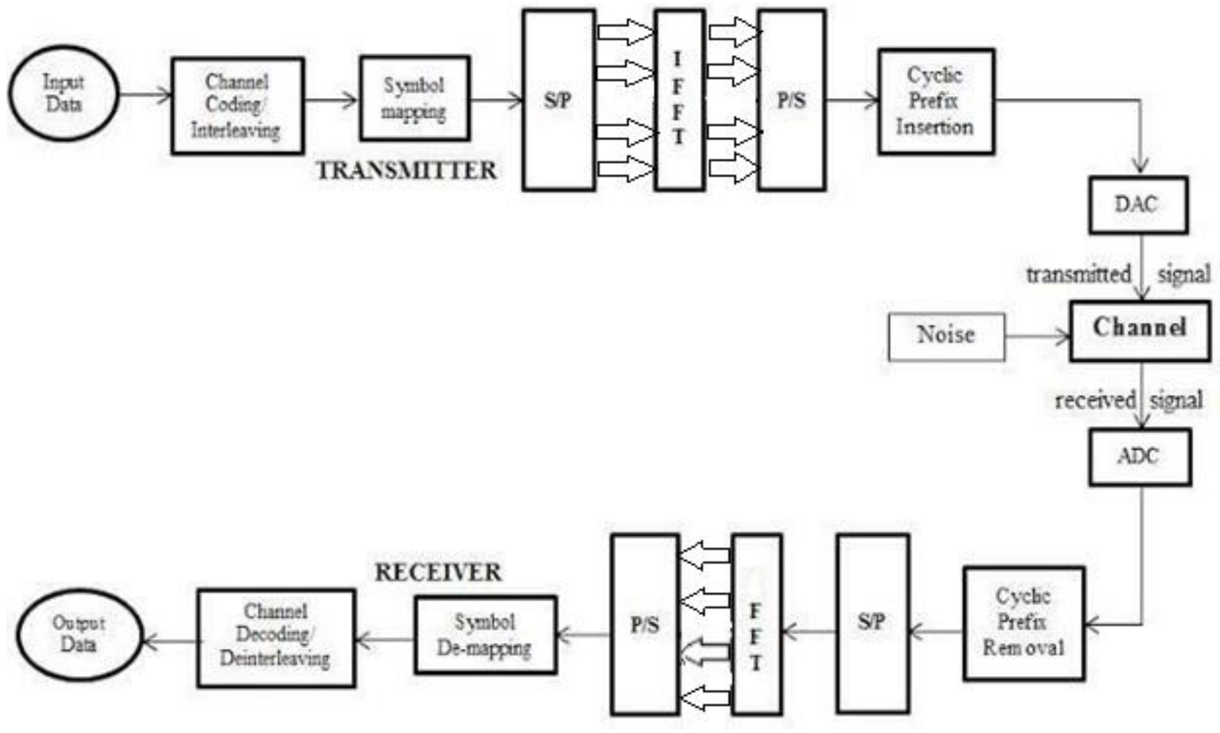
INFERENCE:

1. Thus, different modulation scheme is used for different SNR for lossless transmission of the signal.
2. When the SNR is low, BER is high and hence Throughput is low. Error detection is easier and hence lower order modulation techniques are used.
3. When the SNR is high, BER is low and hence Throughput is high. Error detection becomes difficult and hence higher order modulation techniques are used.

RESULT:

Thus, Adaptive Modulation has been implemented and analyzed using MATLAB

BLOCK DIAGRAM:



EXP NO: 02 **DATE:** 21/02/2024	**Orthogonal Frequency Division Multiplexing**

AIM:

To implement multicarrier modulation technique (OFDM) using MATLAB.

SOFTWARE REQUIRED:

- MATLAB 2022b

THEORY:

- Orthogonal Frequency Division Multiplexing (OFDM) is a digital multi-carrier modulation scheme that extends the concept of single subcarrier modulation by using multiple subcarriers within the same single channel. Rather than transmit a high-rate stream of data with a single subcarrier, OFDM makes use of a large number of closely spaced orthogonal subcarriers that are transmitted in parallel.

- Each subcarrier is modulated with a conventional digital modulation scheme (such as QPSK, 16QAM, etc.) at low symbol rate. However, the combination of many subcarriers enables data rates similar to conventional single-carrier modulation schemes within equivalent bandwidths.

- The OFDM signal can be described as a set of closely spaced FDM subcarriers. In the frequency domain, each transmitted subcarrier results in a sinc function spectrum with side lobes that produce overlapping spectra between subcarriers. This results in subcarrier interference except at orthogonally spaced frequencies.

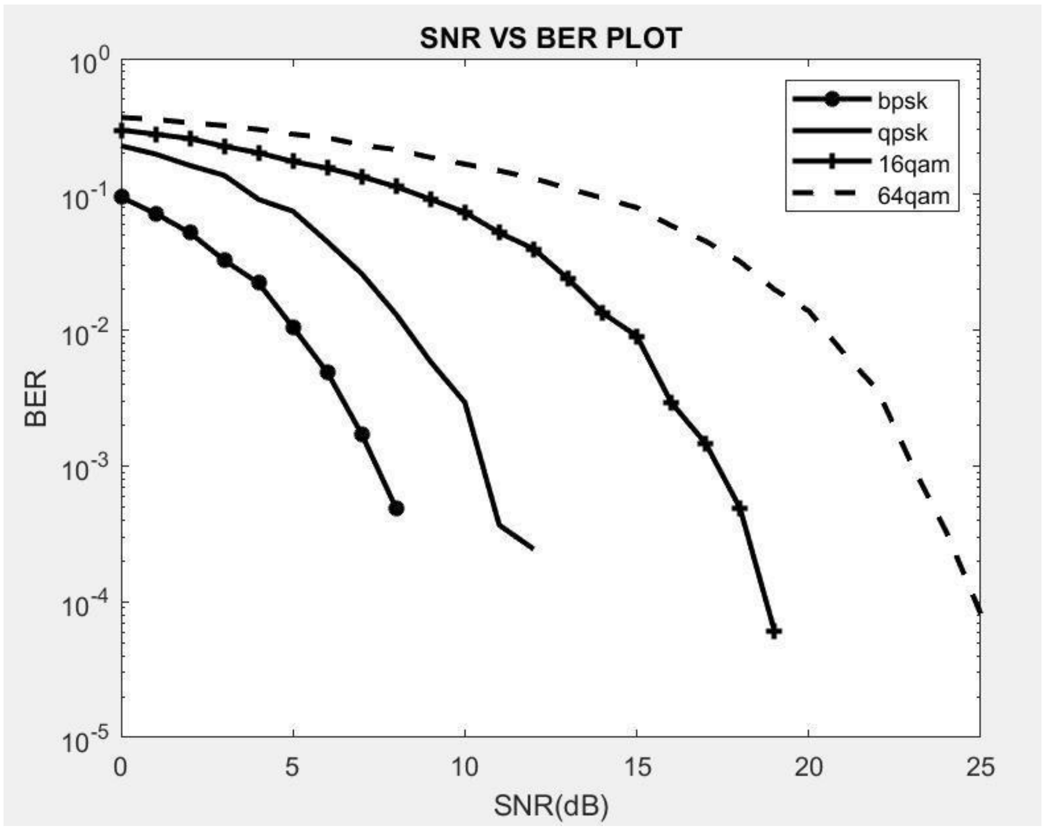
- At orthogonal frequencies, the individual peaks of subcarriers all line up with the nulls of the other subcarriers. This overlap of spectral energy does not interfere with the system’s ability to recover the original signal. The receiver multiplies (i.e., correlates) the incoming signal by the known set of sinusoids to recover the original set of bits sent.

- The use of orthogonal subcarriers allows more subcarriers per bandwidth resulting in an increase in spectral efficiency. In a perfect OFDM signal, Orthogonality prevents interference between overlapping carriers.

ALGORITHM:

1. Generate the data points.
2. Modulate the data.
3. Convert columns to row(serial to parallel)
4. Take IFFT for data in the matrix.
5. Add cyclic prefix to create actual OFDM block.
6. Connect rows to column (parallel to serial).
7. Transmit the OFDM Signal.
8. Received signal is converted to parallel.
9. FFT is performed after removing cyclic prefix.
10. Convert the data to serial stream.
11. Demodulate the data.
12. Repeat it for different SNRs
13. Plot the BER vs SNR graph

MATLAB OUTPUT:



MATLAB CODE:

```
clc; clear close all rng(0) %generating same bit each and every time
n=4096; snr=0:30; biterror=[];
for  i=1:4
if i==1 k=2;
x=randi([0 1],1,n); d=x;
y=pskmod(x,k);
elseif i==2 %QPSK
k=4;
x=randi([0 3],1,n);
b=de2bi(x,'left-msb');
c=b.'; d=reshape(c,1,2*n);
y=pskmod(x,k); %input of pskmod should be in decimal
elseif i==3 %16-qam
x=randi([0 15],1,n);
k=16;
b=de2bi(x,'left-msb'); c=b.';
d=reshape(c,1,4*n); %for biterror comparison
y=qammod(x,k);
elseif i==4 %64-qam
x=randi([0 63],1,n);
k=64; b=de2bi(x,'left-msb');
c=b.'; d=reshape(c,1,6*n);
y=qammod(x,k);
end
p=reshape(y,64,length(y)/64);%max no. of subcarriers
q=ifft(p,64);
s=reshape(q,1,length(y));%serial converter be=[];
for j=0:1:30
h=1./sqrt(rand(1,length(y))+i.*sqrt(rand(1,length(y))));
r=h.*s;t=awgn(r,j,'measured');
m=t./h; %flat fading->hx+n
p11=reshape(m,64,length(y)/64); q11=fft(p11,64);
s11=reshape(q11,1,length(y));
if i==1 %BPSK
y11=pskdemod(s11,k); y14=y11;
elseif i==2 %QPSK
y11=pskdemod(s11,k);
y12=de2bi(y11,'left-msb');
y13=y12.';
y14=reshape(y13,1,2*length(y11));
elseif i==3 %16-qam
y11=qamdemod(s11,k);
y12=de2bi(y11,'left-msb'); y13=y12.';
y14=reshape(y13,1,4*length(y11));
elseif i==4 %64-qam
y11=qamdemod(s11,k);
y12=de2bi(y11,'left-msb'); y13=y12.';
y14=reshape(y13,1,6*length(y11));
end
[num1,e1]=biterr(y14,d); be=[be e1];
end
biterror(i,:)=be;
end
semilogy(snr,biterror(1,:),'-*k','linewidth',2);hold on;
semilogy(snr,biterror(2,:),'-k','linewidth',2);hold on;
semilogy(snr,biterror(3,:),'-+k','linewidth',2);hold on;
semilogy(snr,biterror(4,:),'--k','linewidth',2);hold on;
xlabel('SNR(dB)');  ylabel('BER'); title('SNR VS BER PLOT');
legend('bpsk','qpsk','16qam','64qam')
```

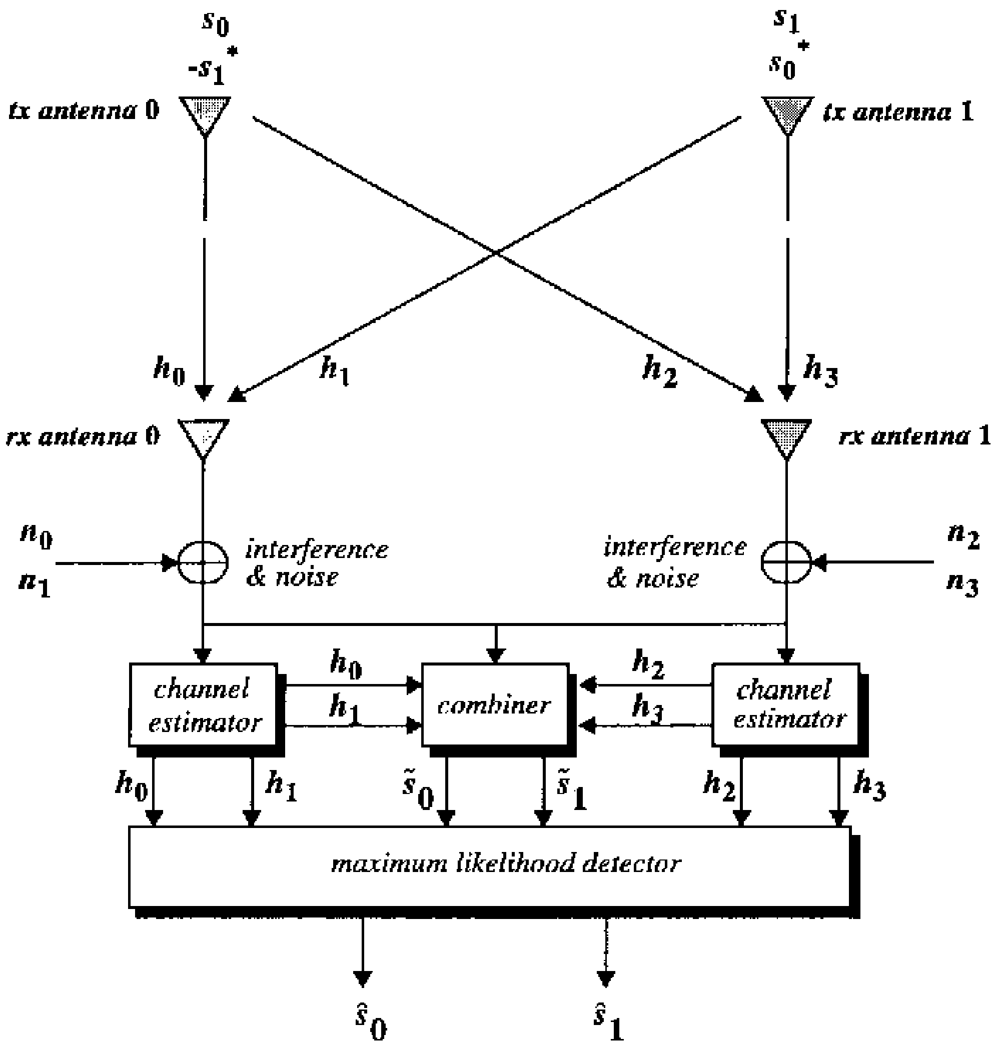
INFERENCE:

The main advantage of OFDM over single carrier schemes is its ability to cope with severe channel conditions (for example, attenuation at high frequency in long carrier wire, narrowband interference and frequency-selective fading due to multipath) without complex equalization filters.

RESULT:

Thus, implementation of multicarrier modulation (Orthogonal Frequency Division Multiplexing) is carried out using MATLAB.

BLOCK DIAGRAM:



EXP NO: 03 **DATE:** 06/03/2024	**Space Time Block Codes**

AIM:

To perform space time block codes (Alamouti coding) for a system that has two transmitter and one receiver.

SOFTWARE REQUIRED:

- MATLAB 2022b

THEORY:

Space-time block codes are used for MIMO systems to enable the transmission of multiple copies of a data stream across a number of antennas and to exploit the various received versions of the data to improve the reliability of data-transfer. Space-time coding combines all the copies of the received signal in an optimal way to extract as much information from each of them as possible.

A space time block code is usually represented by a matrix. Each row represents a time slot and each column represents one antenna's transmissions over time. Space-time block codes (STBC) are a generalized version of Alamouti scheme. These schemes have the same key features. Therefore, these codes are orthogonal and can achieve full transmit diversity specified by the number of transmit antennas. In another word, spacetime block codes are a complex version of Alamouti's space- time code, where the encoding and decoding schemes are the same as there in the Alamouti space-time code in both the transmitter and receiver sides.

At the receiver side, when signals are received, they are first combined and then sent to the maximum likelihood detector where the decision rules are applied.

ALGORITHM:

1. Generate random binary sequence of +1s and -1s.
2. Group them into pairs of two symbols.
3. Code it as per the Alamouti Space Time Code, multiply the symbols with the channels and then add AWGN.
4. Equalize the received symbols.
5. Perform hard decision decoding and count the bit errors.
6. Repeat for multiple values and plot the simulation.

Receiver with Alamouti STBC

In the first time slot, the received signal is,

$$y_1 = h_1x_1 + h_2x_2 + n_1 = [h_1 \ \ h_2]\begin{bmatrix} x_1 \\ x_2 \end{bmatrix} + n_1.$$

In the second time slot, the received signal is,

$$y_2 = -h_1x_2^* + h_2x_1^* + n_2 = [h_1 \ \ h_2]\begin{bmatrix} -x_2^* \\ x_1^* \end{bmatrix} + n_2.$$

where

y_1, y_2 is the received symbol on the first and second time slot respectively, h_1 is the channel from 1^{st} transmit antenna to receive antenna, h_2 is the channel from 2^{nd} transmit antenna to receive antenna, x_1, $x_2$are the transmitted symbols and n_1, n_2 is the noise on 1^{st},2^{nd} time slots.

the above equation can be represented in matrix notation as follows:

$$\begin{bmatrix} y_1 \\ y_2^* \end{bmatrix} = \underbrace{\begin{bmatrix} h_1 & h_2 \\ h_2^* & -h_1^* \end{bmatrix}}\begin{bmatrix} x_1 \\ x_2 \end{bmatrix} + \begin{bmatrix} n_1 \\ n_2^* \end{bmatrix}.$$

Let us define $\boldsymbol{H} = \begin{bmatrix} h_1 & h_2 \\ h_2^* & -h_1^* \end{bmatrix}$

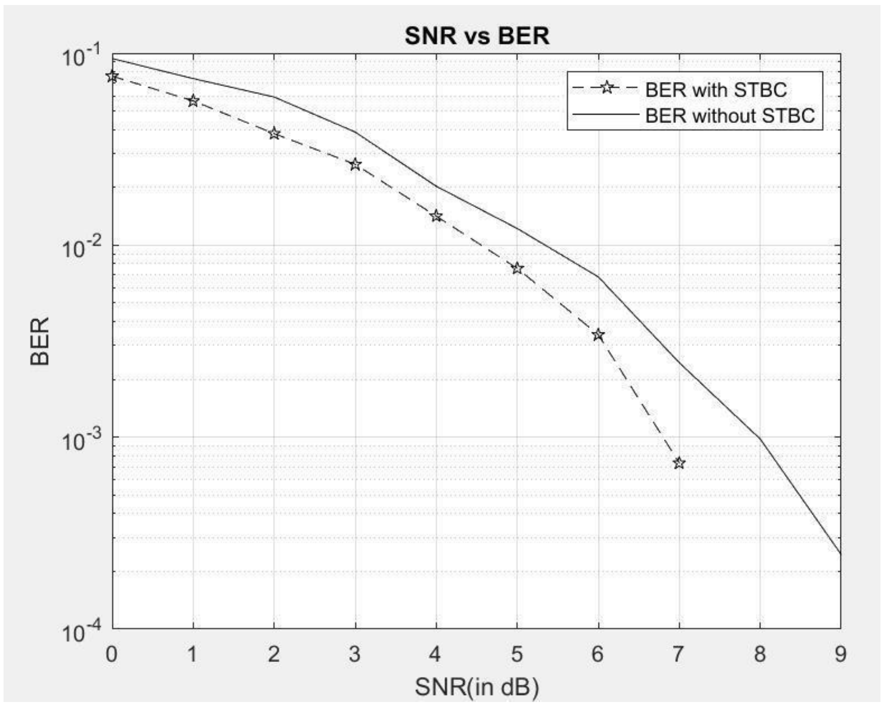
To solve for $\begin{bmatrix} x_1 \\ x_2 \end{bmatrix}$, we know that we need to find the inverse of $\boldsymbol{H}$.

$$(H^H H)^{-1} = \begin{bmatrix} \frac{1}{|h_1|^2+|h_2|^2} & 0 \\ 0 & \frac{1}{|h_1|^2+|h_2|^2} \end{bmatrix}$$

The estimate of the transmitted symbol is,

$$\begin{aligned} \widehat{\begin{bmatrix} x_1 \\ x_2 \end{bmatrix}} &= (H^H H)^{-1} H^H \begin{bmatrix} y_1 \\ y_2^* \end{bmatrix} \\ &= (H^H H)^{-1} H^H \left(H \begin{bmatrix} x_1 \\ x_2 \end{bmatrix} + \begin{bmatrix} n_1 \\ n_2^* \end{bmatrix} \right) \\ &= \begin{bmatrix} x_1 \\ x_2 \end{bmatrix} + (H^H H)^{-1} H^H \begin{bmatrix} n_1 \\ n_2^* \end{bmatrix} \end{aligned}$$

MATLAB OUTPUT:



MATLAB CODE:

```
clc; clear all; close all;
n = randi([0,1],1,4096);
a = reshape(n,length(n),1);
bpskmod = pskmod(a,2);
snr = 0:1:30; h1 = 2+1j; h2 = 1-2j;
y = []; M= []; Op = [];
OpwoutSTBC = []; q = 1;
for l  =  1:length(snr)
for p = 1:2:length((n))-1
c1 = (h1*bpskmod(p,1))+(h2*bpskmod(p+1,1));
M(p,q) = awgn(c1,snr(l),'measured');
c2 = (-h1*conj(bpskmod(p+1,1)))+(h2*conj(bpskmod(p,1)));
M(p+1,q) = awgn(c2,snr(l),'measured');
end
for r = 1:2:(length(n) -1 )
y(r,q) = (conj(h1)*M(r,1)) + h2*conj(M(r+1,1));
y(r+1,q) = (conj(h2)*M(r,1)) - h1*conj(M(r+1,1));
end
t1 = pskdemod(y,2);
Op(l,:) = reshape (t1,1,length(n));
[number,ratio] = biterr(Op,n);
end
semilogy(snr,ratio,'pentagram--C','Color','black')
xlabel("SNR(in dB"); ylabel("BER"); hold on
%without STBC
for l = 1:length(snr)
rec = awgn((h1*bpskmod),snr(l),"measured");
demod = pskdemod(rec,2);
OpwoutSTBC(l,:) = reshape(demod,1,length(n));
[number1,ratio1] = biterr(OpwoutSTBC,n);
end
semilogy(snr,ratio1,'-k')
grid on
legend('BER with STBC','BER without STBC')
xlabel("SNR(in dB"); ylabel("BER");
title("SNR vs BER");
```

RESULT:

Thus, the Space Time Block Code using Alamouti coding scheme is studied, and the Bit Error Rate is analyzed.

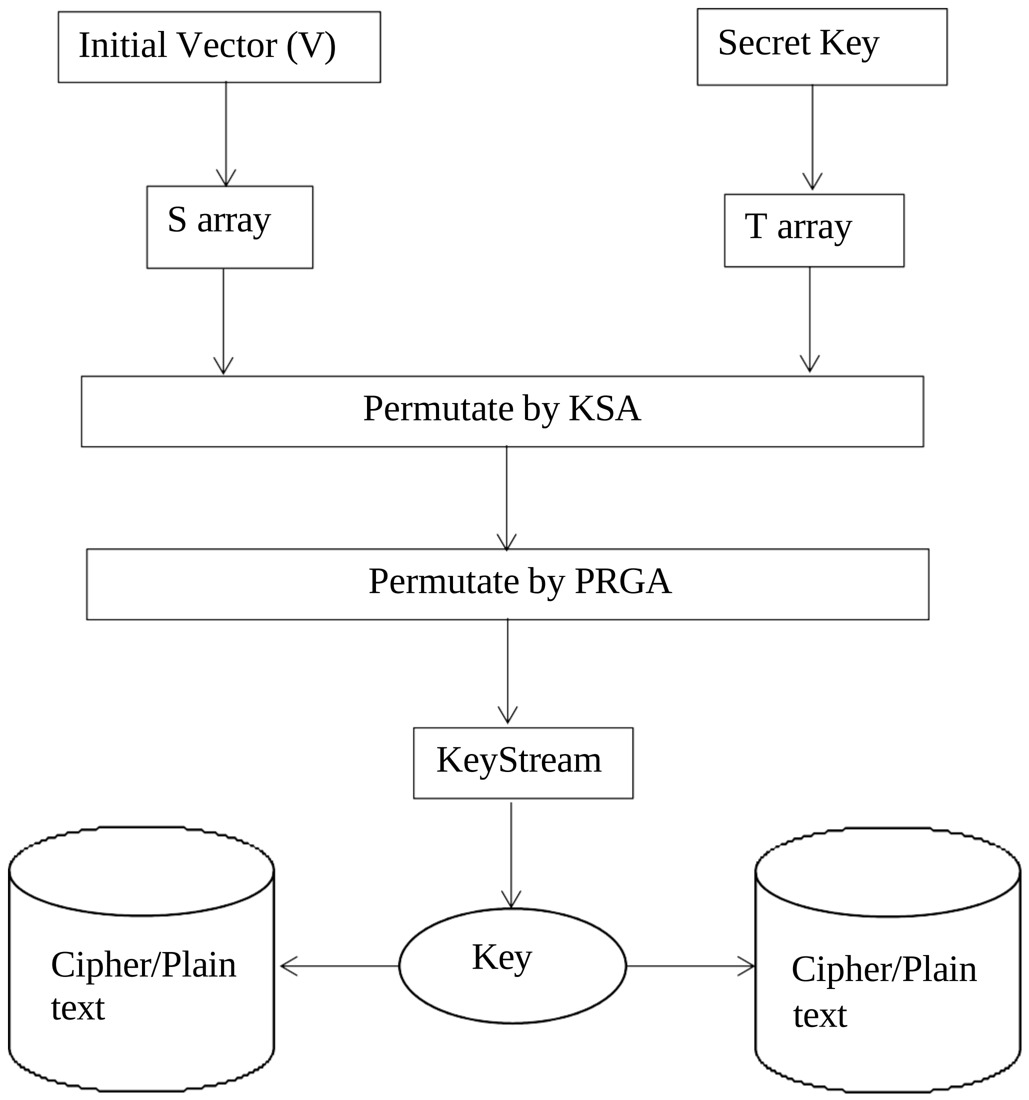
INFERENCE:

1. No need of Channel information since the transmitter antenna is independent of Channel State
2. Since it requires two Transmit antennas, this scheme consumes more power.

CONCLUSION:

The Bit Error Rate is getting reduced while using STBC coding.

BLOCK DIAGRAM:



EXP NO: 04 **DATE:** 20/03/2024	**Implementation of Network Security Protocols – RC4 Encryption Algorithm**

AIM

To implement RC4 stream cipher encryption and decryption in MATLAB.

MATERIALS REQUIRED:

MATLAB R2022b

THEORY:

RC4 means Rivest Cipher 4 invented by Ron Rivest in 1987 for RSA Security. It is a Stream Ciphers. Stream Ciphers operate on a stream of data byte by byte. RC4 stream cipher is one of the most widely used stream ciphers because of its simplicity and speed of operation. It is a variable key-size stream cipher with byte-oriented operations. It is generally used in applications such as Secure Socket Layer (SSL), Transport Layer Security (TLS).

For encryption:

The user enters the Plaintext and a secret key.
For the secret key entered, the encryption engine creates the keystream using the
KSA and PRGA algorithms.
Plaintext is XORed with the generated keystream. Because RC4 is a stream cipher,
byte-by-byte XORing is used to generate the encrypted text.
This encrypted text is now sent in encrypted form to the intended recipient.

For Decryption:

The same byte-wise X-OR technique is used on the ciphertext to decrypt it.

ALGORITHM:

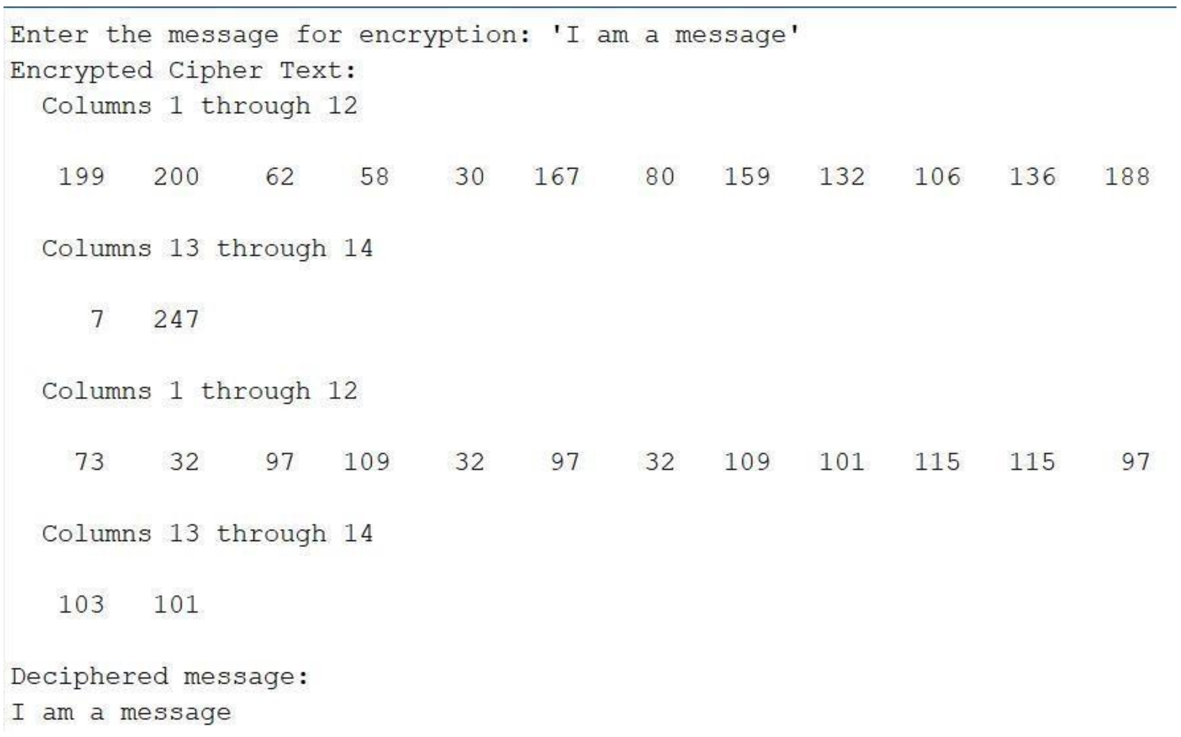
KEY SCHEDULING ALGORITHM

```
Initialization
for i = 0 to 255 do
S[i] = i;
T[i] = K[i mod K – len];
j = 0;
for i = 0 to 255 do
        j = (j +S[i] + T[i]) mod 256;
        Swap (S[i], S[j]);
```

PSEUDO RANDOM GENERATION ALGORITHM (STREAM GENERATION)

```
 i, j = 0;
 while (true)
 i = (i + 1) mod 256;
 j = (j + S[i]) mod 256;
 Swap (S[i], S[j]);
T = (S[i] + S[j]) mod 256;
 k = S[t];
```

MATLAB OUTPUT:



```
Enter the message for encryption: 'I am a message'
Encrypted Cipher Text:
  Columns 1 through 12

   199   200    62    58    30   167    80   159   132   106   136   188

  Columns 13 through 14

     7   247

  Columns 1 through 12

    73    32    97   109    32    97    32   109   101   115   115    97

  Columns 13 through 14

   103   101

Deciphered message:
I am a message
```

MATLAB CODE:

```
clc; clear all; close all;
s = zeros(1,256);
for i = 1:256
s(i)  =  i;
end
msg_1 = input('Enter the message for encryption: ');
message = [];
message = [message double(char(msg_1))];
key = [2 4 6 8 2 1 6 1];
t = zeros(1,256);
for i = 1:256
t(i) = mod(i,length(key));
end
%KSA
i = 0; j =0;
for i = 1:256
j = mod(j+s(i)+t(i),256) + 1;
s([i j]) = s([j i]);
j = j-1;
end
%PRGA
i = 0; j = 0; k = [];
for m = 1:length(message)
i = mod(i+1,256);
j = mod(j + s(i),256);
s([i j]) = s([j i]);
k = [k s(mod(s(i) + s(j),256))];
end
encrypted_message = [];
for i = 1:length(message)
encrypted_message = [encrypted_message bitxor(message(i),k(i))];
end
disp("Encrypted Cipher Text: "); disp(char(encrypted_message));
decrypted_message = [];
for i = 1:length(message)
decrypted_message = [decrypted_message bitxor(encrypted_message(i),k(i))];
end
disp(decrypted_message)
disp('Deciphered message: '); disp(char(decrypted_message))
```

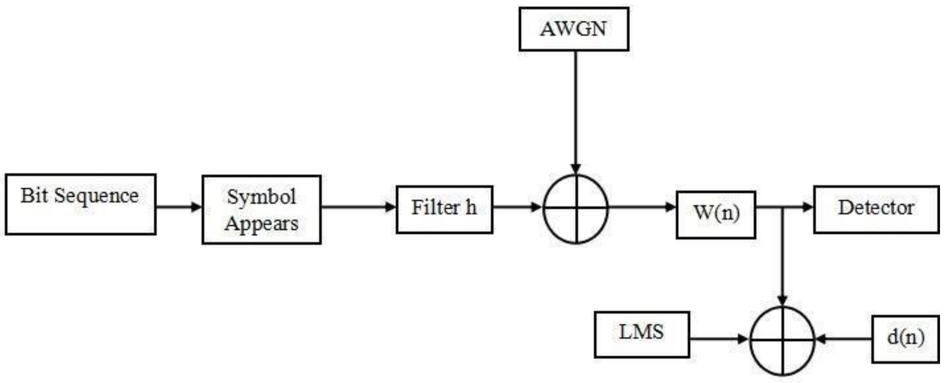
INFERENCE

RC4 is a stream cipher that is simple to use and the speed of operationis also higher. It does not require a lot of memory and can also be implemented on large streams of data. But, RC4 is vulnerable and insecure at times due to which very few applications use it nowadays.

RESULT:

Thus RC4 Encryption stream cipher algorithm is implemented using MATLAB.

BLOCK DIAGRAM:



EXP NO: 05 **DATE:** 28/02/2024	**Performance of Equalizers**

AIM
:

To implement equalization techniques for wireless channels using Least Mean Squares (LMS) and Zero Forcing algorithms

SOFTWARE REQUIRED:

MATLAB 2022b

THEORY:

A channel equalizer is an important component of a communication system and is used to mitigate the ISI (inter symbol interference) introduced by the channel. The equalizer depends upon the channel characteristics. These are usually employed to reduce the depth and duration of the fades experienced by a receiver in a local area which are due to motion. An equalizer within a receiver compensates for the average range of expected channel amplitude and delay characteristics. Equalizers must be adaptive since the channel is generally unknown and varies with time.

Least Mean Squares (**LMS**) algorithms are a class of adaptive filter used to mimic a desired filter by finding the filter coefficients that relate to producing the least mean square of the error signal (difference between the desired and the actual signal). It is a stochastic gradient descent method in that the filter is only adapted based on the error at the current time.

Zero Forcing Equalizers is a linear equalizer which uses unit impulse response to compensate for channel effect. An overall impulse response is equal to one for the defected symbol and zero for all other received symbols. The noise may be increased in this process.

_LEAST MEAN SQUARES (LMS) ALGORITHM

ALGORITHM FORMULATION:

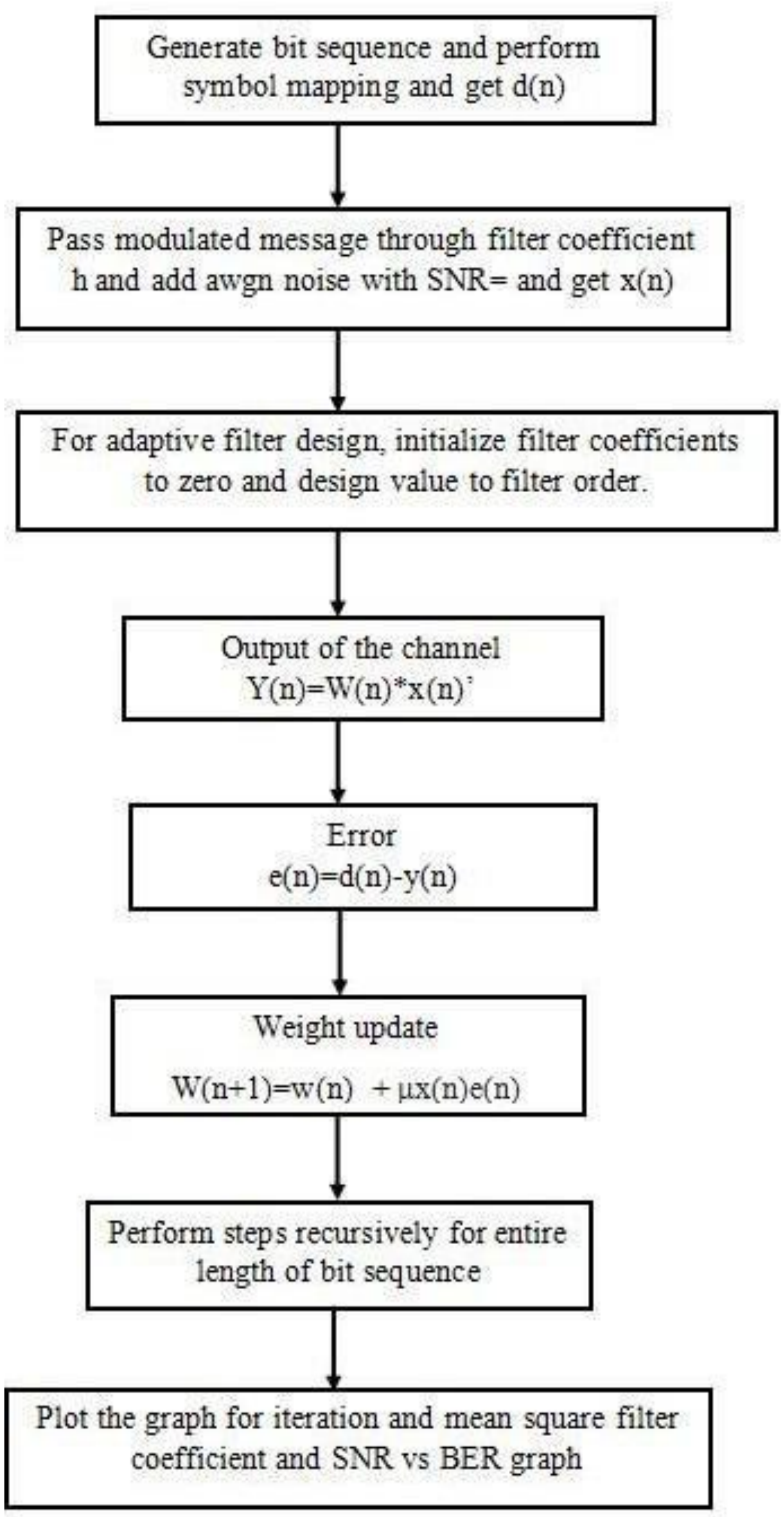
From the method of steepest descent, the weight vector equation is given by,

$$w(n+1) = w(n) + \frac{1}{2}\mu(E(e^{2}(n))$$

where,

- $w(n)$ → weight update vector matrix on the nth instant
- μ → step-size parameter and controls the convergence characteristics of the LMS algorithm

FLOWCHART: (LMS)


Generate bit sequence and perform symbol mapping and get d(n)
Pass modulated message through filter coefficient h and add awgn noise with SNR= and get x(n)
For adaptive filter design, initialize filter coefficients to zero and design value to filter order.
Output of the channel Y(n)=W(n)*x(n)'
Error e(n)=d(n)-y(n)
Weight update W(n+1)=w(n) + µx(n)e(n)
Perform steps recursively for entire length of bit sequence
Plot the graph for iteration and mean square filter coefficient and SNR vs BER graph

- $e^2(n)$ → mean square error(at the nth instant) between the beamformer output y(n) and the reference signal *d(n)* which is given by,

$$e^2(n) = [d(n) - w^H x(n)]2$$

where,

  - w^H → Hermitian of weight update vector matrix
  - x(n)→Input sequence

- $\nabla(E\{e^2()\})$→ Gradient performed on the expectation of the mean square error at the nth instant

In the method of steepest descent, the biggest problem is the computation involved in finding the values r and R matrices in real time. The LMS algorithm on the other hand simplifies this by using the instantaneous values of covariance matrices r and R instead of their actual values i.e.

$$R(n) = x(n)x^H(n)$$

$$r(n) = d(n)x(n)$$

Therefore, the weight update can be given by the following equation,

$$w(n+1) = w(n) + \mu x(n)[d(n) - x^H(n)w(n)]$$

$$w(n+1) = w(n) + \mu x(n)e(n)$$

The LMS algorithm is initiated with an arbitrary value *w(0)* for the weight vector at *n=0*. The successive corrections of the weight vector eventually lead to the minimum value of the mean squared error.

Therefore, the LMS algorithm can be summarized in following equations;

$$\text{Output} \rightarrow \mathrm{y}(n) = w^H x(n)$$

$$Error \rightarrow e(n) = d(n) - y(n)$$

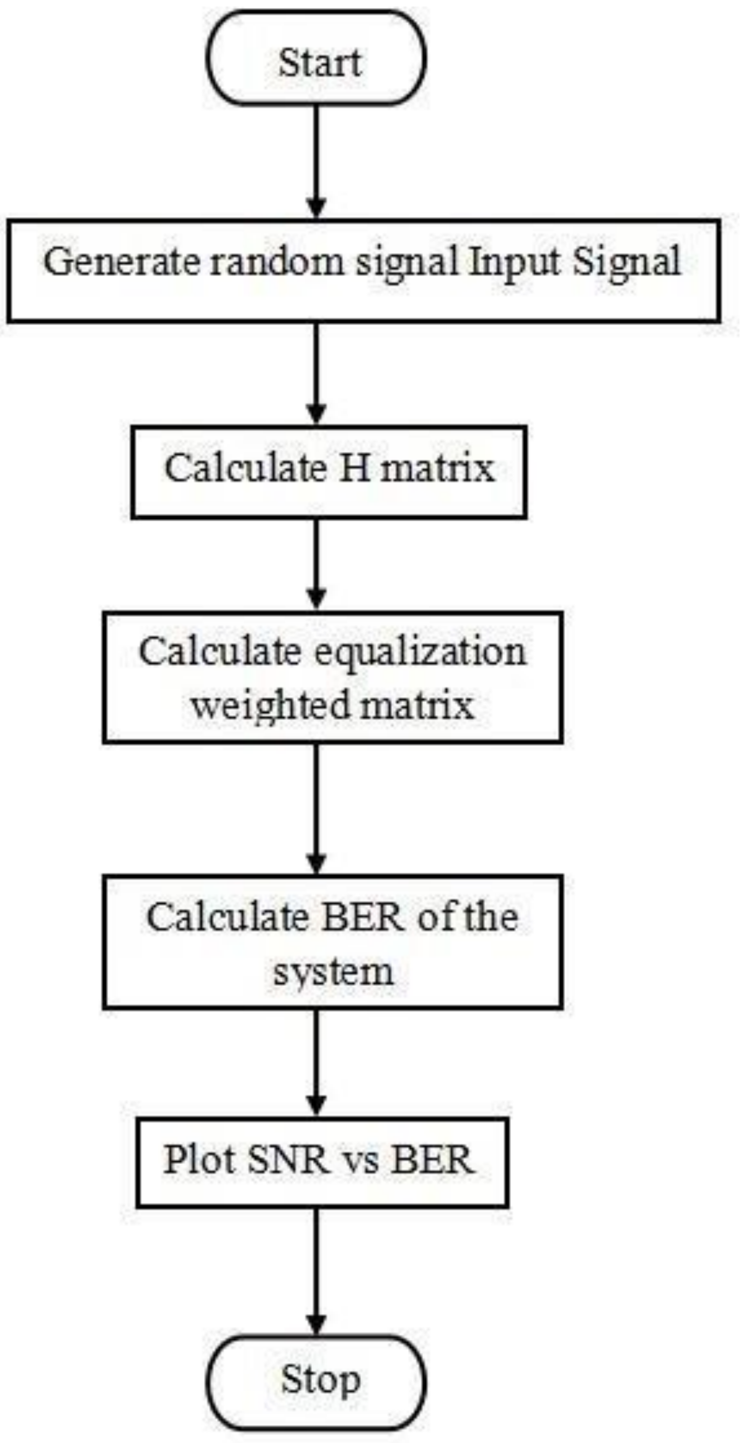
$$Weight \rightarrow w(n+1) = w(n) + \mu x(n)e(n)$$

ZERO FORCING EQUALIZERS

ALGORITHM FORMULATION:

The following equations are used to formulate the algorithm used in Zero Forcing Equalizers:

FLOWCHART: (Zero forcing Equalizers)


Start
Generate random signal Input Signal
Calculate H matrix
Calculate equalization weighted matrix
Calculate BER of the system
Plot SNR vs BER
Stop

$$y_k = h_0 x_k + h_1 x_{k-1} + n_0$$

$$y_{k+2} = h_0 x_{k+2} + h_1 x_{k+1} + n_2$$

$$y_{k+1} = h_0 x_{k+1} + h_1 x_k + n_1$$

$$\begin{bmatrix} y_{k+2} \\ y_{k+1} \\ y_k \end{bmatrix} = \begin{bmatrix} h_0 & h_1 & 0 & 0 \\ 0 & h_0 & h_1 & 0 \\ 0 & 0 & h_0 & h_1 \end{bmatrix} . \begin{bmatrix} x_{k+2} \\ x_{k+1} \\ x_k \\ x_{k-1} \end{bmatrix} + \begin{bmatrix} n_2 \\ n_1 \\ n_0 \end{bmatrix}$$

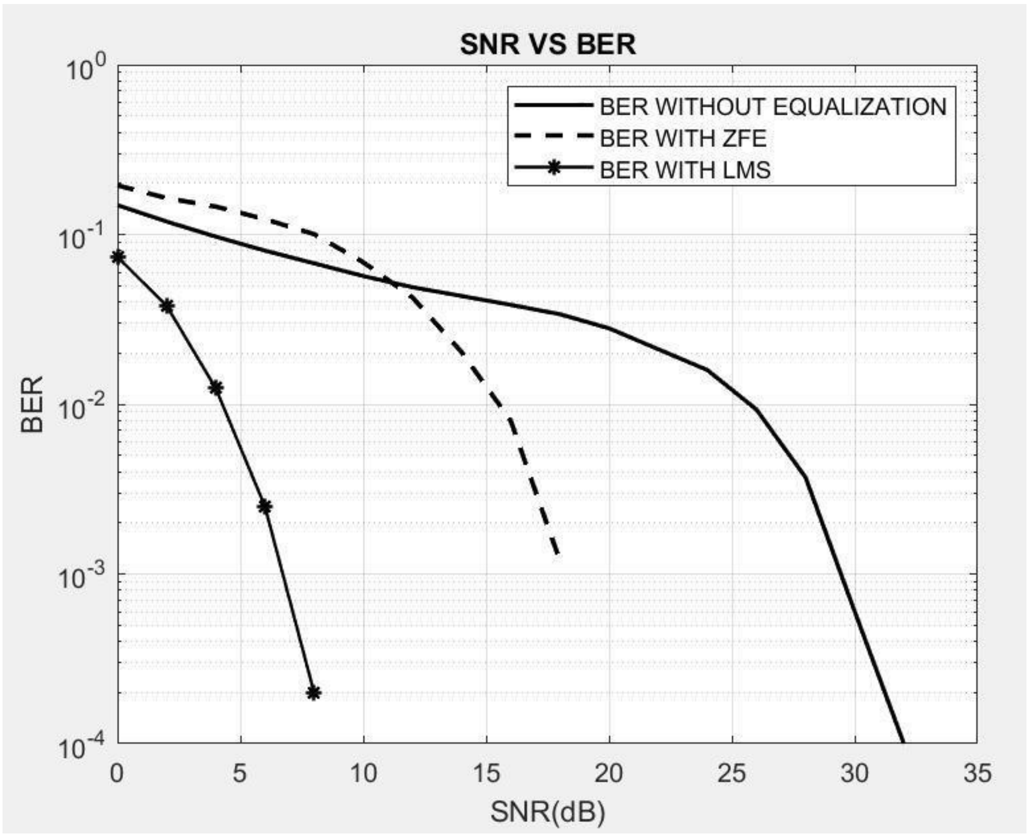
$$\overline{1_2} = C^T H = \begin{bmatrix} 0 & 0 & 1 & 0 \end{bmatrix}$$

$$C = (HH^T)^{-1} . \overline{1_2}$$

where,

- → Output sequence at instance *k*
- → Input sequence at instant *k*
- → Noise sequence at instant *k*
- h_0 h_1 → Equalizing filter coefficients
- → Zero Forcing Constant

MATLAB OUTPUT:



MATLAB CODE:

```
clc; clear all; close all;
message = randi([0,1],1,10000);
snr = 0:2:40; mod = 2;
L = 2; r = 3; % 3 TAP EQUALISER
modulated_bpsk_msg = pskmod(message,mod);
h0 = 1; h1 = 0.7;
H = zeros(r,r+L-1); % ORDER OF H MATRIX
% FORMING H MATRIX FOR 3 TAP EQUALISER
for p = 1:r
H(p,p:p+L-1) = [h0 h1];
end
x = []; % FORMING INPUT MATRIX X
X = modulated_bpsk_msg; X_1 = circshift(X,1);
X_1(1) = 0; X1 = circshift(X,-1);
X1(end) = 0; X2 = circshift(X,-2);
X2(end-1:end)= 0; x = [X2;X1;X;X_1];
C = ((H*H')\H)*[0;0;1;0];
ber_without_ZFE = []; ber_with_ZFE = [];
for p = 1:length(snr)
y = awgn([h0 h1]*[X;X_1],snr(p),'measured');
Noise = y - ([h0 h1]*[X;X_1]);
Noise_1 = circshift(Noise,-1);
Noise_1(end) = 0; Noise_2 = circshift(Noise,-2);
Noise_1(end-1:end) = 0;
Y = (H*x)+ [Noise_2;Noise_1;Noise]; X_PRIME = C.'*Y;
Demodulated_BPSK_msg_with_ZFE =pskdemod(X_PRIME',mod);
Demodulated_BPSK_msg_without_ZFE  = pskdemod(y',mod);
[number1,ratio1] = biterr(message,Demodulated_BPSK_msg_with_ZFE');
[number2,ratio2] = biterr(message,Demodulated_BPSK_msg_without_ZFE');
ber_with_ZFE = [ber_with_ZFE,ratio1];
ber_without_ZFE = [ber_without_ZFE,ratio2];
end
semilogy(snr,ber_with_ZFE,'-k','LineWidth',1.6)
hold on
semilogy(snr,ber_without_ZFE,'--k','LineWidth',1.8)
hold on
n = length(message); M = 25; w = zeros(1,M);
wi = zeros(1,M); E = []; mu = 0.0005;
msg_bpsk = pskmod(message,mod);
ber_without_LMS = []; ber_with_LMS = [];
for k = 1:length(snr)
msg_rx = awgn(msg_bpsk, snr(k), 'measured');
for i = M:n
E(i) = msg_bpsk(i) - wi*msg_rx(i:-1:i-M+1)';
wi = wi + 2*mu*E(i)*msg_rx(i:-1:i-M+1);
end
msg_eq = zeros(n,1);
for i = M:n
j = msg_rx(i:-1:i-M+1);
msg_eq(i) = ((wi)*(j)');
end
Demod_with_LMS = pskdemod(msg_eq,mod)';
Demod_without_LMS = pskdemod(msg_rx,mod);
[n1,r1] = biterr(message,Demod_with_LMS);
[n2,r2] = biterr(message,Demod_without_LMS);
ber_without_LMS = [ber_without_LMS,r1];
ber_with_LMS = [ber_with_LMS,r2];
end
grid on; semilogy(snr,ber_with_LMS,'-*k','LineWidth',1.2)
title('SNR VS  BER'); xlabel('SNR(dB)');ylabel('BER');
legend('BER WITHOUT EQUALIZATION','BER WITH ZFE','BER WITH LMS');
```

INFERENCE:

- In the Least Mean Squares (LMS) algorithm of equalization techniques of wireless channels, when the epoch iterations increase, error decreases thereby decreasing the Mean Squared Error (MSE).
- In the case of Zero Forcing Equalizer, it can be observed that it nullifies inter-symbol interference (ISI) in the modulated signals. But noise obtained in the process increases accordingly.
- Hence, comparing both the algorithms, we can observe that Least Mean Squares (LMS) algorithm is efficient than that of Zero Forcing Equalization technique in terms of noise performance.

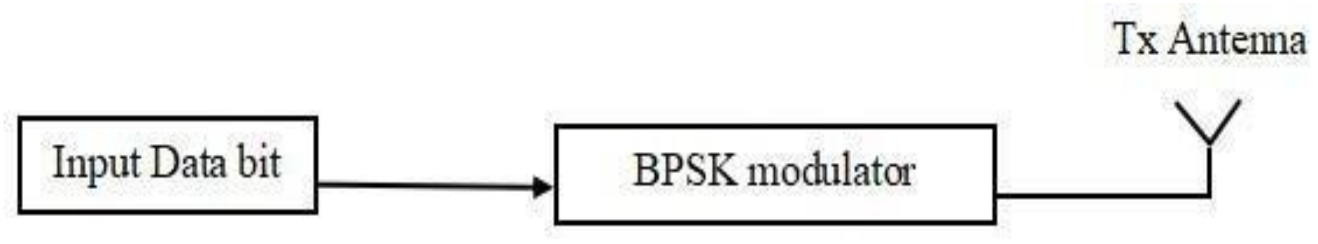
RESULT:

Hence, channel equalization using the Least Mean Squares (LMS) algorithm and Zero Forcing Equalizer are successfully performed using MATLAB and the cost function/BER-SNR graphs for the same are plotted.

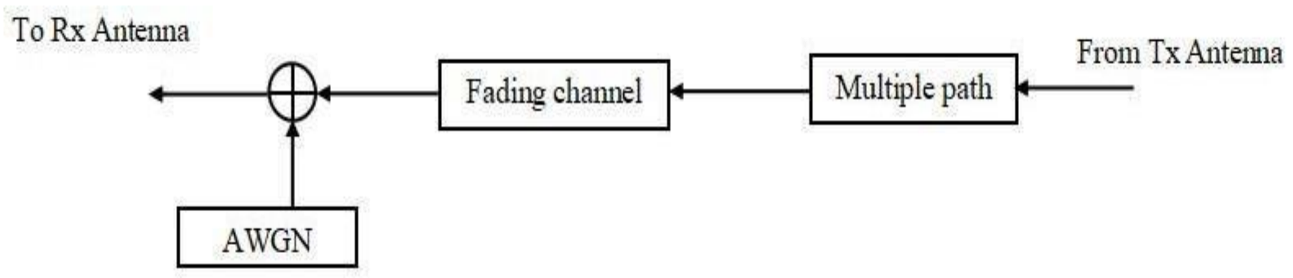
g

BLOCK DIAGRAM:

Transmitter:



Channel:



EXP NO:06 **DATE:** 07/02/24	**Characteristics Of Wireless Channels**

AIM

To analyse BER performance of Rayleigh and Rician channel.

SOFTWARE REQUIRED:

MATLAB R2022b

THEORY:

RICIAN DISTRIBUTION:

A Rician distribution is one way to model the paths scattered signal stake to a receiver. Specifically, this distribution models line-of sight scatter — transmissions between two stations in view of each other that have an unobstructed path between them. Line-of-sight scatter includes FM radio waves, microwaves, MRI images in the presence of noise, and satellite transmissions. The distribution also models Rician fading, which is a way to show how signal cancellations affect radio propagation. The probability density function formula is: RAYLEIGH DISTRIBUTION Rayleigh distribution is a continuous probability distribution for positive-valued random variables. The data can be given by the mean value and a lower bound, or by a parameter and a lower bound. These are interconnected by a well-documented relationship given in the literature. For instance, if the mean μ=2 and the lower bound is γ=0.5, then θ=1.59577 and the standard deviation is σ=1.0454.

The Doppler shift of this wave is given by

$$s(t) = I(t)\cos\omega_c t + Q(t)\sin\omega_c t$$

$$I(t) = \sum_{i=1}^{N} a_i \cos(\omega_{di} t + \Phi_i)$$

$$Q(t) = \sum_{i=1}^{N} a_i \sin(\omega_{di} t + \Phi_i)$$

and the envelope R is given by:

$$R = \sqrt{[I(t)]^2 + [Q(t)]^2}$$

$$\omega_{di} = \frac{\omega_c v}{c}\cos\psi_i$$

RAYLEIGH DISTRIBUTION

Rayleigh distribution is a continuous probability distribution for positive-valued randon variables. The data can be given by the mean value and a lower bound, or by a parameter and a lower bound. These are interconnected by a well-documented relationship given in the literature For instance, if the mean μ=2 and the lower bound is γ=0.5,then θ=1.59577 and the standarc deviation is σ=1.0454.

$$s(t)=\sum_{i=1}^{N-1} a_i \cos\left(\omega_c t+\omega_{di} t+\varphi_i\right) + k_d \cos\left(\omega_c t+\omega_d t\right)$$

$$f(r)=\frac{r}{\sigma^2} e^{-\left\{\frac{r^2+k_d{}^2}{2\sigma^2}\right\} \cdot I_0\left\{\frac{rk_d}{\sigma^2}\right\}}$$

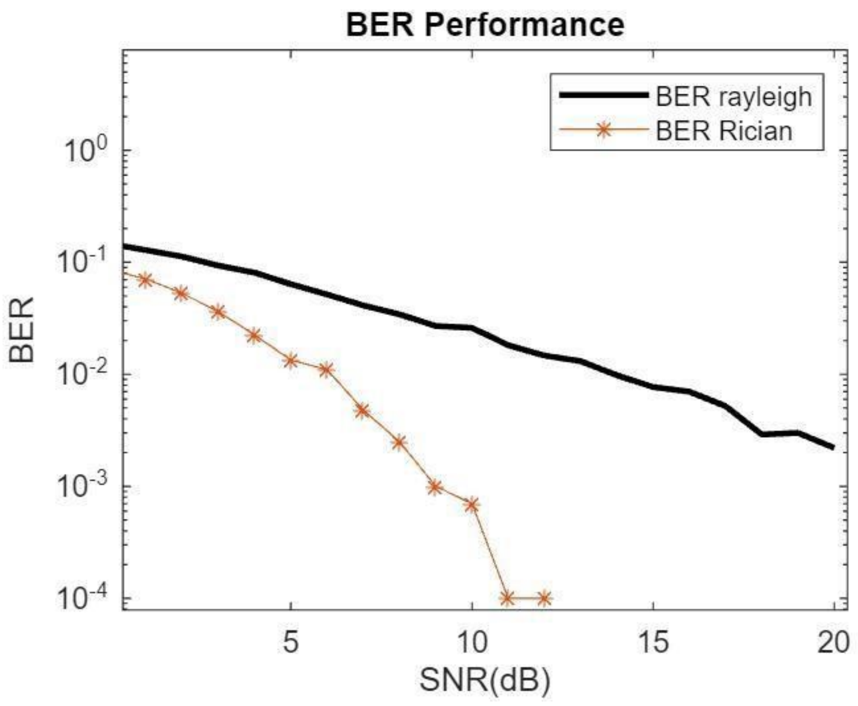
where $I_0(0)$ the 0th order is modified Bessel function of the first kind

ALGORITHM:

BER VS SNR PERFORMANCE OF CHANNELS

1. Generate BPSK modulated wave.
2. Generate noise sequence that are gaussian distributed.
3. Multiply the modulated wave with the Rayleigh coefficient and add the noise.
4. Detect the received signal, compare it with the input signal to find the error.
5. Plot the BER vs SNR graph for Rayleigh channel.
6. Multiply the modulated wave with the rician coefficient and add the noise.
7. Detect the received signal, compare it with the input signal to find the error.
8. Plot the BER vs SNR graph for rician channel.

MATLAB OUTPUT:



MATLAB CODE:

```
%Rician and Rayleigh Channel
clc clear all close all
n = 10000;
i = randi([0,1],1,n);
i1 = 2*i -1;
a = randn(1,n); b = randn(1,n);
rc = 1/sqrt(2)*(sqrt(a.^2+b.^2));
for l = 0:1:20
snr = 10^(l/10); sdev = sqrt(0.5/snr);
N = random('norm',0,sdev,[1,n]);
yrc = rc.*i1+N; YR = (yrc>=0);
ErrorR = sum((xor(YR,i)));
ber_R(l+1) = ErrorR/n;
end
q = 0:1:20;
semilogy(q,ber_R(q+1),'k-','LineWidth',2);
hold on;
axis([0 20 10^-5 1]);
xlabel('SNR(dB)'); ylabel('BER');
k1 = 10; mean = sqrt(k1/(k1+1));
sigma = sqrt(1/(2*(k1+1)));
Nr2 = randn(1,length(i1))*sigma+mean;
Ni2 = randn(1,length(i1))*sigma;
No3 = sqrt(Nr2.^2+Ni2.^2);
for k = 0:1:20
snrl = 10^(k/10); Np = 1/snrl;
sd = sqrt(Np/2); No = random('Normal',0,sd,1,length(i1));
t1 = i1.*No3+No; z1 = t1./No3;
op1 = (z1>0);
Berr(k+1) = sum(xor(op1,i))/n;
end
k = 0:1:20;
semilogy(k,Berr(k+1),'*-');
hold on;
axis([0 20 10^-5 1]);
title(' BER Performance ');
xlabel('SNR(dB)'); ylabel('BER');
legend('BER rayleigh','BER Rician');
```

INFERENCE:

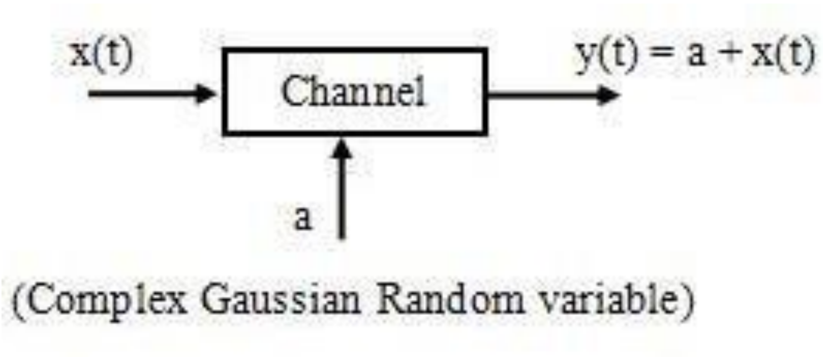
- BER performance of Rayleigh & Rician channel were plotted.
- AWGN and Rician channels provided the best performances in rura areas as compared to urban areas whereas Rayleigh channel provide premium efficiency in urban localities.
- When the Rician factor (k) is equal to zero then it is called Rayleigh channel and when k increases graph tends to be Gaussian PDF.

RESULT:

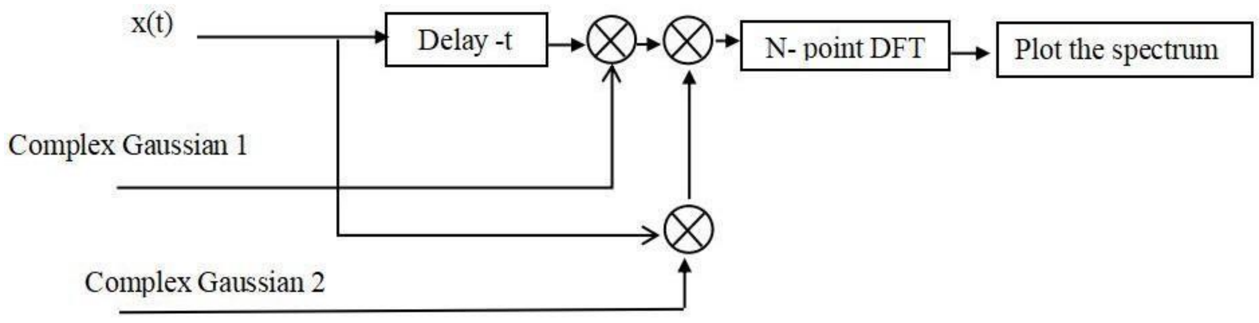
Thus the Rayleigh & Rician channel distribution in wireless channel were analysed and the BER performance was analysed.

BLOCK DIAGRAM:

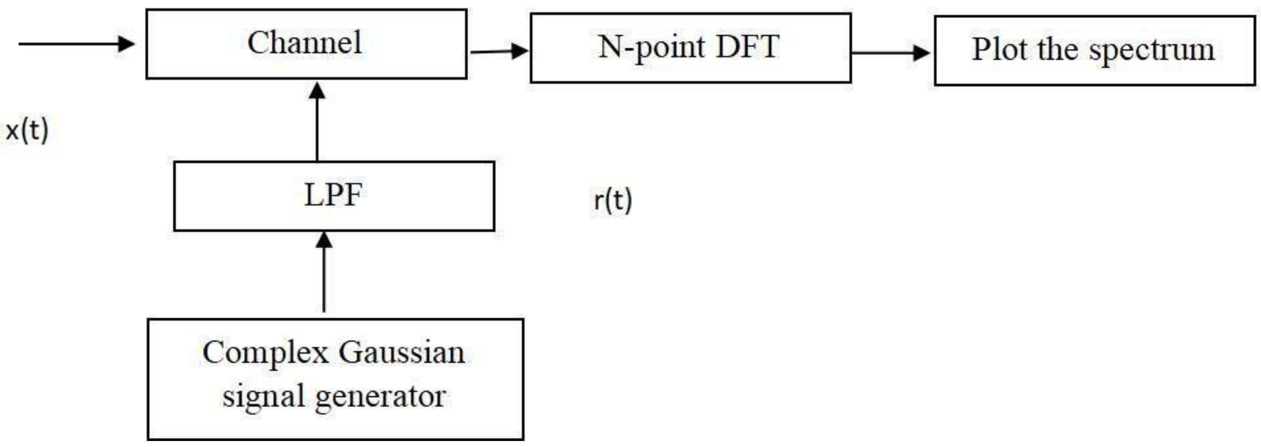
Slow and Flat Fading



Slow and Frequency Selective Fading



Fast and Flat Fading



EXP NO: 07 **DATE:** 21/02/2024	**Simulation And Implementation of types of Fading**

AIM

To plot the spectra of two tone signal due to slow flat fading, slow Frequency selective fading, fast flat fading, fast frequency selective fading.

SOFTWARE REQUIRED:

MATLAB R2022b

THEORY:

Based on multipath delay spread there are two types of small scale fading viz. flat fading and frequency selective fading. These multipath fading types depend on propagation environment.

FLAT FADING

The wireless channel is said to be flat fading if it has constant gain and linear phase response over a bandwidth which is greater than the bandwidth of the transmitted signal. In this type of fading all the frequency components of the received signal fluctuate in same proportions simultaneously. It is also known as non-selective fading

- Signal BW > Channel BW
- Symbol Period > Delay Spread

The effect of flat fading is seen as decrease in SNR. These flat fading channels are known as amplitude varying channels or narrow band channels.

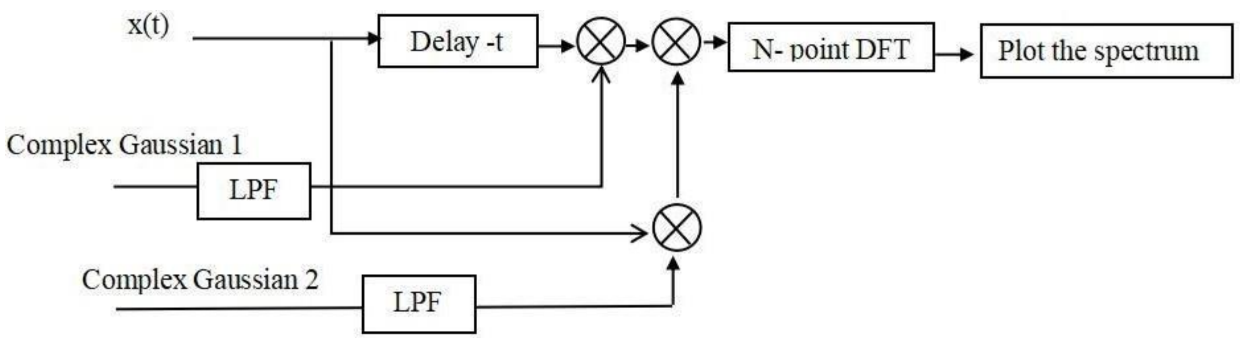
FREQUENCY SELECTIVE FADING

It affects different spectral components of a radio signal with different amplitudes.Hence the name selective fading

- Signal BW> Channel BW
- Symbol period < Delay Spread

Based on doppler spread there are two types of fading viz. fast fading and slow fading. These doppler spread fading types depend on mobile speed i.e.speed of receiver with respect to transmitter.

Fast and Frequency Selective Fading



FAST FADING

The phenomenon of fast fading is represented by rapid fluctuations of signal over small areas (i.e.bandwidth). When the signals arrive from all the directions in the plane, fast fading will be observed for all direction so motion. Fast fading occurs when channel impulse response changes very rapidly within the symbol duration.

- High doppler spread
- Symbol period >Coherence time
- Signal Variation< Channel variation

This parameters result into frequency dispersion or time selective fading due to doppler spreading. Fast fading is result of reflections of local objects and motion of objects relative to those objects.

SLOW FADING

Slow fading is result of shadowing by buildings, hills, mountains and other objects over the path.

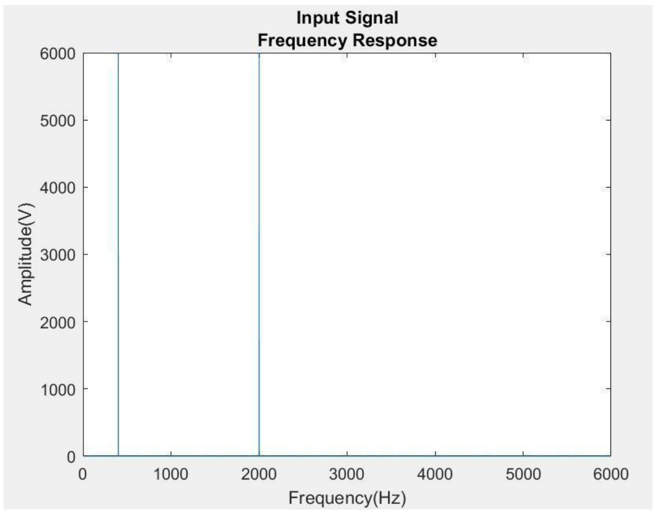
- Low Doppler Spread
- Symbol period << Coherence time
- Signal Variation >Channel Variation

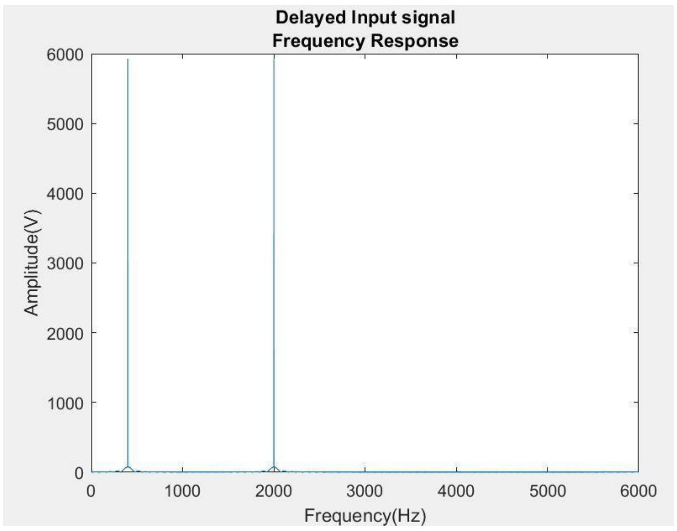
Slow fading results in a loss of SNR. Error correction coding and receiver diversity techniques are used to overcome effects of slow fading.

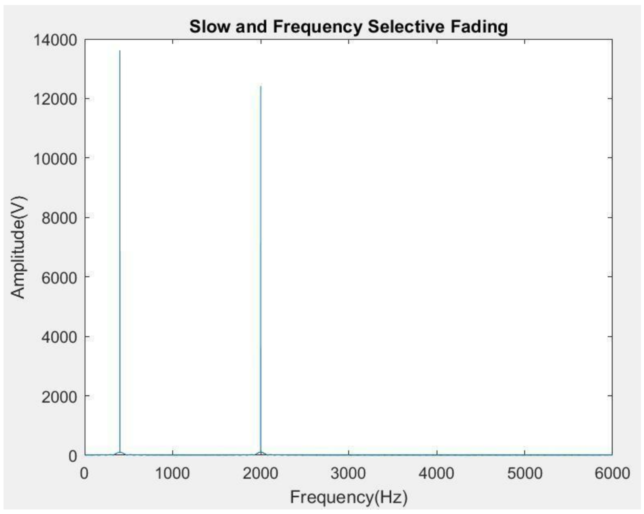
ALGORITHM:

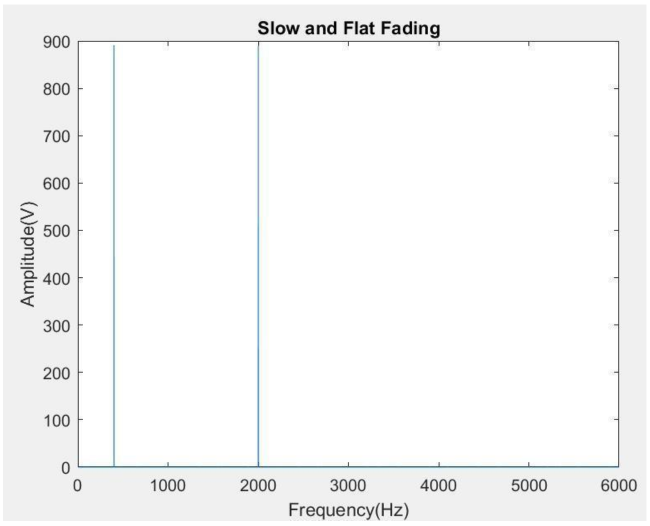
1. Generate a two tone message signal.
2. Fix sampling frequency.
3. Generate a delayed signal.
4. Find the FFT of message signal.
5. For slow & flat fading, generate random complex gaussian variable and multiply with message signal.
6. For slow & frequency selective fading, generate two random complex gaussian variable. Multiply one of the variable with message and the other variable with delayed signal.
7. For fast& flat fading, generate a random complex Gaussian variable and pass it through low pass filter and multiply it with message signal.
8. For fast & frequency selective fading, generate two random complex Gaussian variable. Multiply one of the variable with message and the other variable with delayed signal.Pass it through lowpass filter and multiply it with message signal. Find the FFT of signal and plot it.

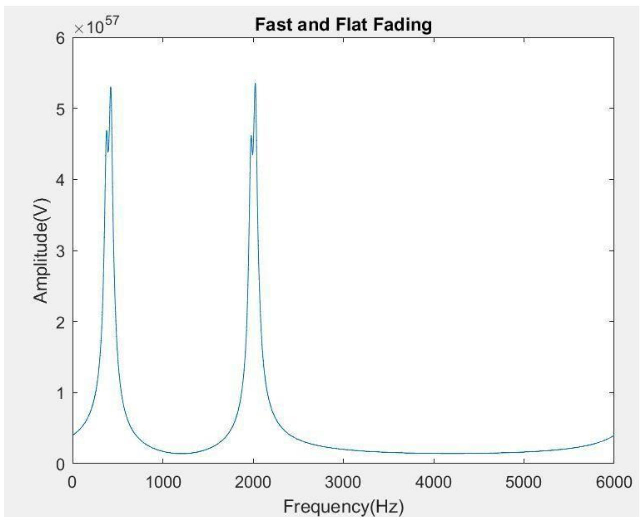
MATLAB OUTPUT:

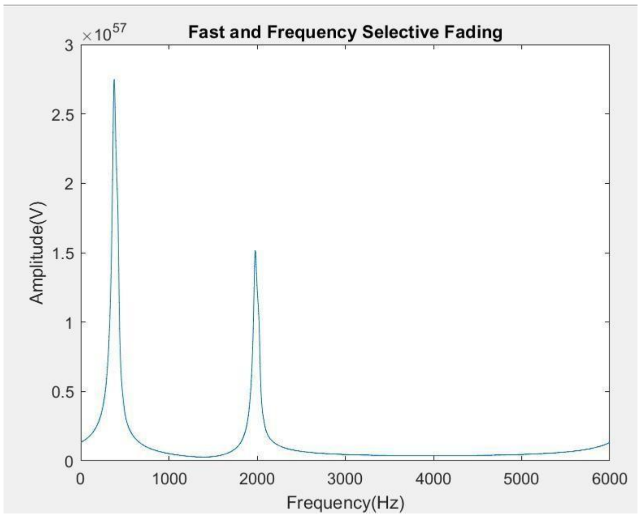












MATLAB CODE:

```
clc, clear all; close all;
fs = 6000; f1 = 2000; f2 = 400;
ts = 0:(1/fs):1-(1/fs);
signal1 = exp(complex(0,2*pi*f1*ts));
signal2 = exp(complex(0,2*pi*f2*ts));
message = signal1+signal2;
delayed = [zeros(1,77) message(1:length(message)-77)];
%FREQUENCY RESPONSE
freqres = fft(message); figure;
magnitudeplot = abs(freqres);
plot(1:fs,magnitudeplot);
title({'Input Signal';'Frequency Response'});
xlabel('Frequency(Hz)'); ylabel('Amplitude(V)');
figure; freqres2=fft(delayed);
plot(1:fs,abs(freqres2));
title({'Delayed Input signal';'Frequency Response'});
xlabel('Frequency(Hz)');
ylabel('Amplitude(V)');
%Slow and Flat Fading
h = randn + (1i*randn);
y1 = message.*h;
freqres = fft(y1);
magnitudeplot = abs(fft(y1));
figure;
plot(1:fs,magnitudeplot(1:fs));
title('Slow and Flat Fading');
xlabel('Frequency(Hz)'); ylabel('Amplitude(V)');
%Slow and Frequency Selective Fading
h1 = randn + (1i*randn);
h2 = randn + (1i*randn);
trans2 = (h1.*message) + (h2.*delayed);
magnitudeplot = abs(fft(trans2));
figure; plot(1:fs,magnitudeplot);
title('Slow and Frequency Selective Fading');
xlabel('Frequency(Hz)'); ylabel('Amplitude(V)');
%Fast and Flat Fading
fd = 10;%doppler frequency
hs = randi([0 1],1,length(ts)) +(1i*randi([0 1],1,length(ts)));
[b,a] = butter(12,((2*fd)/1000));
lpf = filter(b,a,hs);
trans3 = lpf.*message;
magnitudeplot = abs(fft(trans3));
figure; plot(1:fs,magnitudeplot);
title('Fast and Flat Fading');
xlabel('Frequency(Hz)'); ylabel('Amplitude(V)');
% Fast and Frequency Selective Fading
ht = randi([0 1],1,length(ts)) +1i*(randi([0 1],1,length(ts)));
lpf1 = filter(b,a,ht); trans4 = (lpf1.*message) +(lpf1.*delayed);
magnitudeplot = abs(fft(trans4));
figure;
plot(1:fs,magnitudeplot);
title('Fast and Frequency Selective Fading');
xlabel('Frequency(Hz)'); ylabel('Amplitude(V)');
```

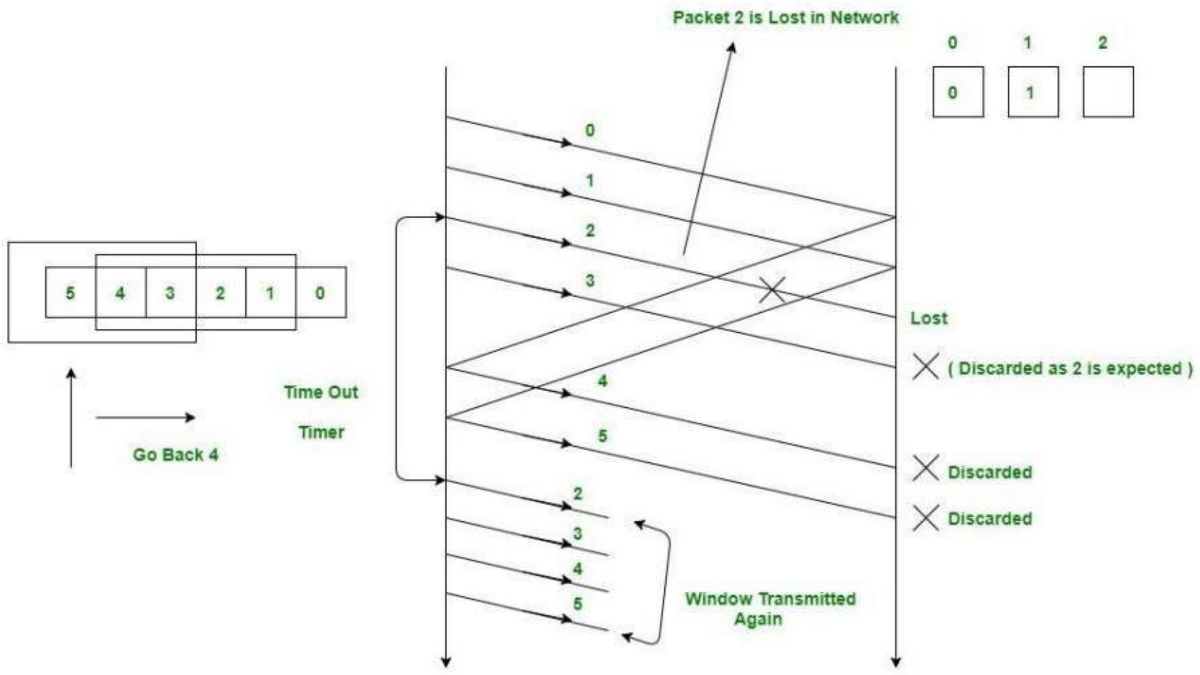
INFERENCE

- Amplitude remains the same in slow flat fading.
- Amplitude varies for slow and frequency selective fading.
- Amplitude remains same but signal is spread in fast and flat fading.
- Amplitude changes and the signal is spread over the frequencies in flat and frequency selective fading.

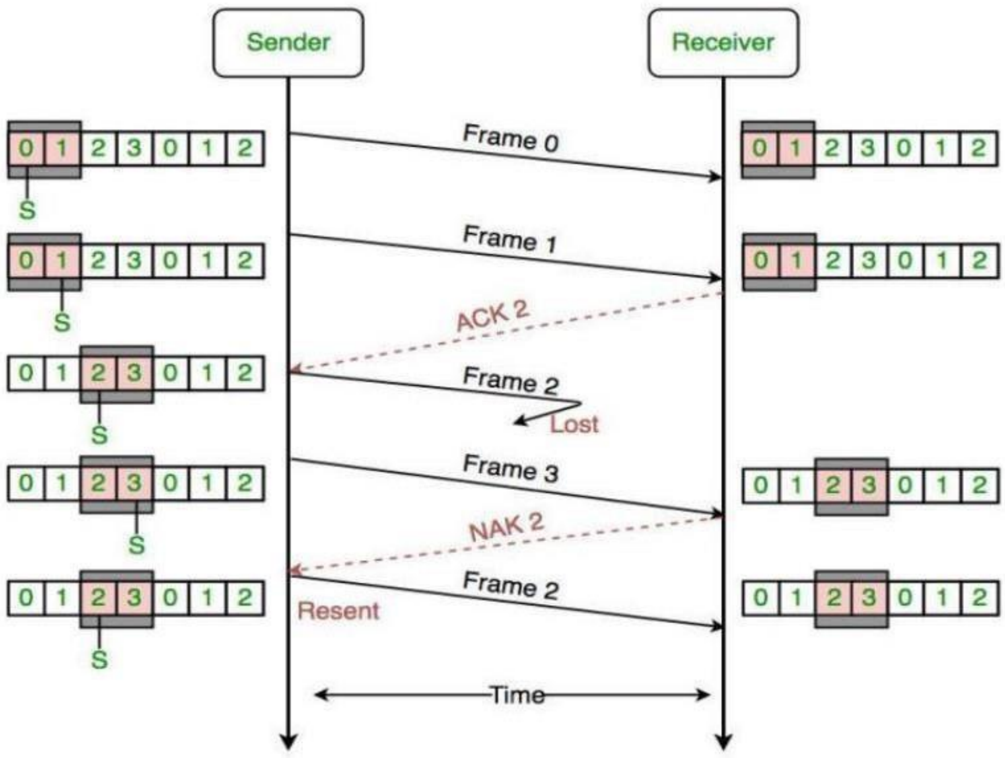
RESULT

Thus we have simulated and plotted the spectra of four different types of fading were analysed using MATLAB and verified successfully.

GO BACK N protocol:



SELECTIVE REPEAT protocol:



EXP NO: 08 **DATE: 07/02/24**	**LLC Protocols**

AIM:

To implement

- Go Back 'N' Protocol
- Selective Repeat Protocol

and to determine the throughput and delay for each.

SOFTWARE REQUIRED:

NETSIM Software

THEORY:

GO BACK 'N' PROTOCOL:

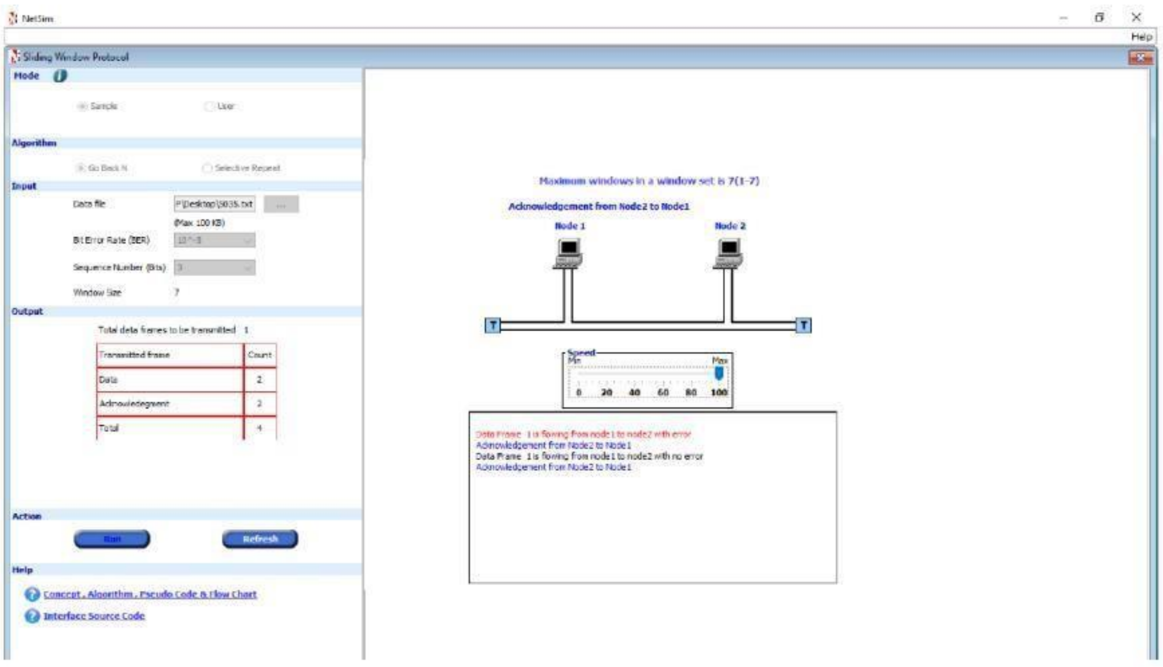
Go Back 'N' is a connection-oriented transmission. The sender transmits the frames continuously. Each frame in the buffer has a sequence number starting from 1 and increasing up to the window size. The sender has a window i.e., a buffer to store the frames. This buffer size is the number of frames to be transmitted continuously. The size of the window depends on the protocol designer

SELECTIVE REPEAT PROTOCOL:

It is similar to Go Back 'N' Protocol, but the sender send frame only after the reception of ACK signal. It may be used as a protocol for delivery and ACK for message units for delivery of subdivided message. It is used as a protocol for delivery of message sender continuous to send frames specifies by windows size even after becoming frameless. Once the sender has sent all the frame in its windows, it resends the frame number given by ACK and continuous where it left off.

GO BACK ‘N’ PROTOCOL

LAYOUT:



Mode: (●) Sample ( ) User

Algorithm: (●) Go Back N ( ) Selective Repeat

Input

Data file: F:\Desktop\5035.txt (Max 100 KB)

Bit Error Rate (BER): 10^-5

Sequence Number (Bits): 3

Window Size: 7

Output

Total data frames to be transmitted 1

Transmitted frame	Count
Data	2
Acknowledgement	2
Total	4

Maximum windows in a window set is 7(1-7)

Acknowledgement from Node2 to Node1

```
Data Frame  1 is flowing from node1 to node2 with error
Acknowledgement from Node2 to Node1
Data Frame  1 is flowing from node1 to node2 with no error
Acknowledgement from Node2 to Node1
```

TABLE: (Sequence number : 3 bits)

BER	TOTAL DATA FRAMES TO BE TRANSMITTED	DATA	ACKNOWLEDGEMENT	TOTAL
10^-5	1	2	2	4
10^-6	1	2	2	4
10^-7	1	1	1	2
10^-8	1	1	1	2
10^-9	1	1	1	2
NO ERROR	1	1	1	2

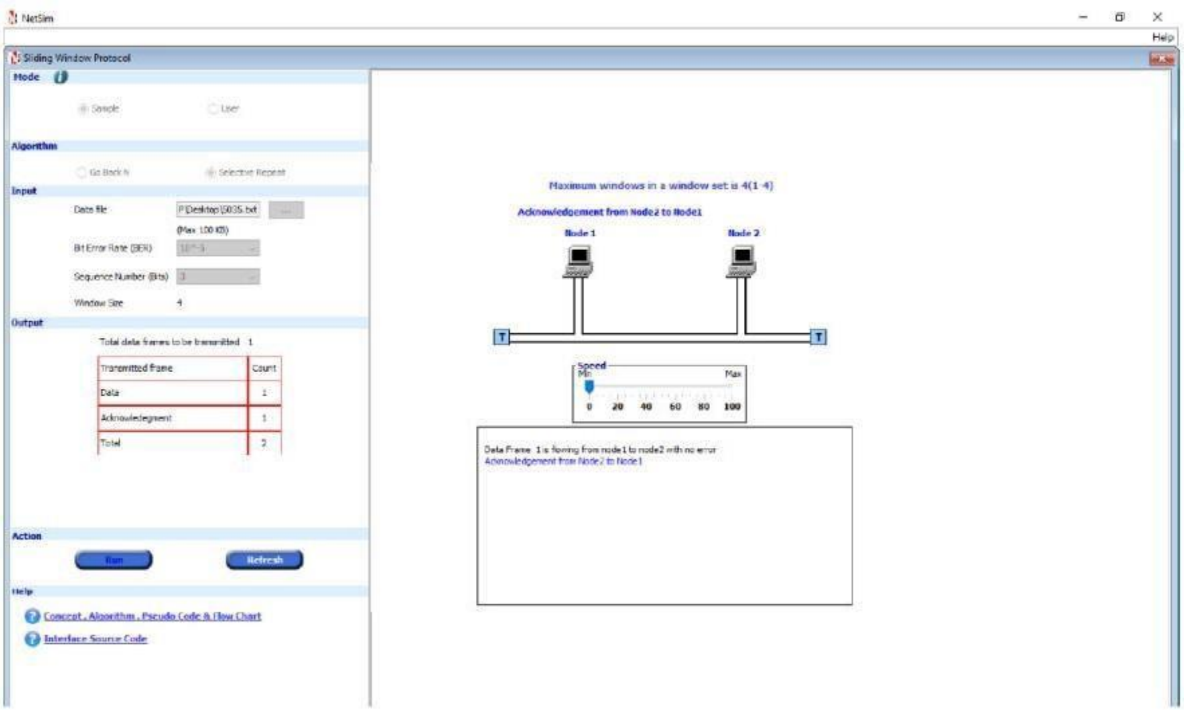
ALGORITHM:

GO BACK 'N' PROTOCOL:

- The source code transmits the frames continuously.
- Each frame in the buffer has a sequence number starting from 1 and increasing up to the window size.
- The source code has a window i.e., a buffer to store the frames. This buffer size is the number of frames to be transmitted continuously. The size of the window depends on the protocol designer.
- For the first frame, the receiving node forms a positive acknowledgement if the frame is received without any error.
- If subsequent frames are received without error (up to window size) cumulative positive acknowledgement is formed.
- If the subsequent frame is received with error, the cumulative acknowledgement error-free frames are transmitted. If, In the same window two frames or more frames are received with error, the second and the subsequent error frames are neglected. Similarly, even the frames received without error after the receipt of a frame with error are neglected.
- The source code re-transmits all frames of window from the first error frame.
- If the frames are errorless in the next transmission and if the acknowledgement is error-free, the window slides by the number of error- free frames being transmitted.
- If the acknowledgement is transmitted with error, all the frames of window at source are re-transmitted and window doesn't slide.
- This concept of replacing the transmission from the first error frame in the window is called as Go Back-N transmission flow control protocol.

SELECTIVE REPEAT PROTOCOL

LAYOUT:



Data file: P:\Desktop\5035.txt (Max 100 KB)
Bit Error Rate (BER): 10^-5
Sequence Number (Bits): 3
Window Size: 4

Total data frames to be transmitted: 1

Transmitted frame	Count
Data	1
Acknowledgement	1
Total	2

Maximum windows in a window set is 4(1-4)

Acknowledgement from Node2 to Node1

Data Frame 1 is flowing from node1 to node2 with no error
Acknowledgement from Node2 to Node1

TABLE: (Sequence number : 3 bits)

BER	TOTAL DATA FRAMES TO BE TRANSMITTED	DATA	ACKNOWLEDGEMENT	TOTAL
10^-5	1	1	1	2
10^-6	1	1	1	2
10^-7	1	1	1	2
10^-8	1	1	1	2
10^-9	1	1	1	2
NO ERROR	1	1	1	2

SELECTIVE REPEAT PROTOCOL:

- The source node transmits the frames continuously.
- Each frame in the buffer has a sequence number starting from 1 and increasing up to the window size.
- The source node has a window i.e., buffer to store the frames. This buffer size is the number of frames to be transmitted continuously.
- The receiver has a buffer to store the received frames. The size of the buffer depends upon the window size defined by the protocol designer.
- The source node transmits frames continuously till the window size is exhausted. If any of the frames are received with error only those frames are requested for retransmission (with a negative acknowledgement).
- If all the frames are received without error, a cumulative positive acknowledgement is sent.
- If there is an error in frame 3, an acknowledgement for the frame 2 is sent and then only frame 3 is retransmitted. Now the window slides to get the next frames to the window.
- If acknowledgement is transmitted with error, all the frames of window are retransmitted. Else ordinary window sliding takes place. (*In implementation part, Acknowledgement error is not considered).
- If all the frames transmitted are errorless the next transmission is carried out for the new window.
- This concept of repeating the transmission for the error frames only is called Selective Repeat transmission flow control protocol.

PROCEDURE:

1. Open NETSIM.
2. Click Programming >> Transmission Flow Control.
3. Select Sample.
4. Select Go Back-N transmission.
5. Enter input data and BER.
6. Click Link to execute the program.
7. Repeat steps 1 to 3.
8. Select Selective Repeat protocol.
9. Enter input data and BER. Click Link to execute the program.

INFERENCE:

It is found that Selective Repeat Protocol is the most optimum out of these two LLC protocols. But the complexity equipment’s are high. So, a trade off exists the total number of transmissions taken place and complexity.

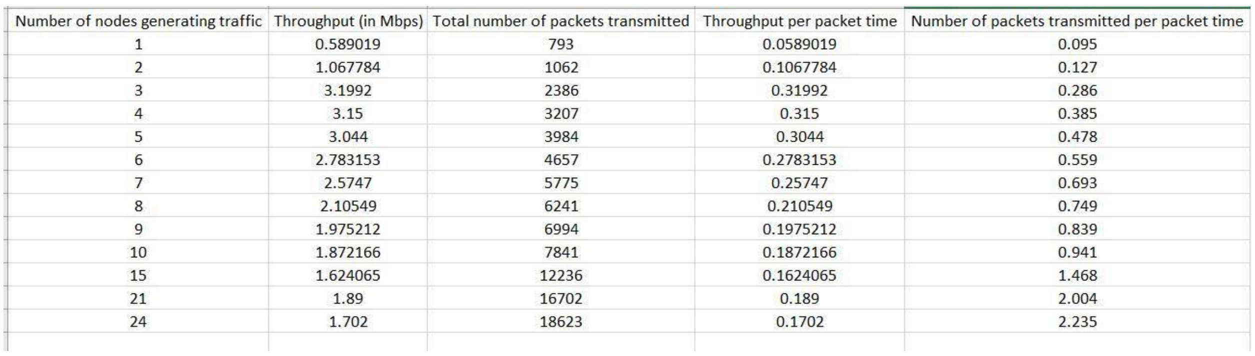
RESULT:

Thus, the LLC protocols such as “Go Back ‘N’” and “Selective Repeat” were studied using NETSIM and their performance were analysed.

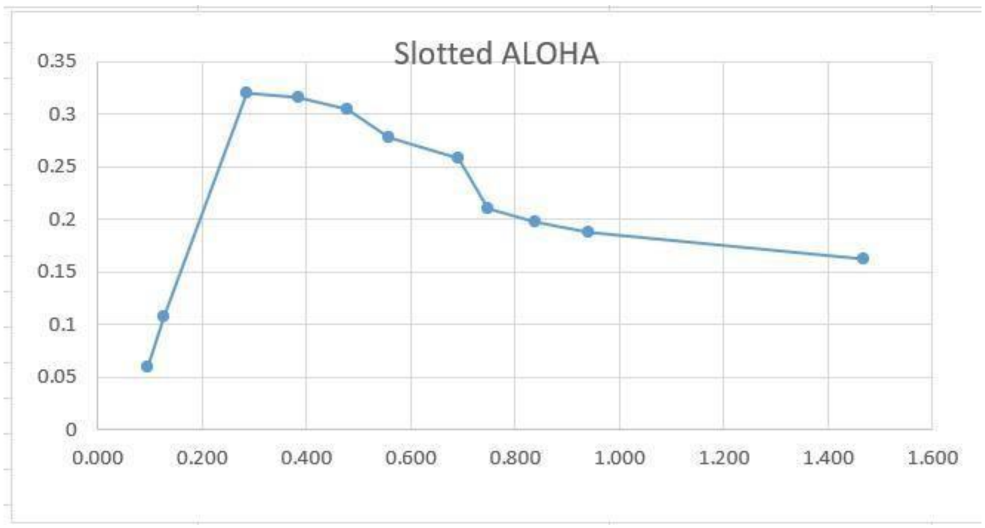
CONCLUSION:

Both the LLC protocols were studied and it can be concluded that Selective Repeat is more effective but also more complex as compared to Go back n protocol.

SLOTTED ALOHA:



Number of nodes generating traffic	Throughput (in Mbps)	Total number of packets transmitted	Throughput per packet time	Number of packets transmitted per packet time
1	0.589019	793	0.0589019	0.095
2	1.067784	1062	0.1067784	0.127
3	3.1992	2386	0.31992	0.286
4	3.15	3207	0.315	0.385
5	3.044	3984	0.3044	0.478
6	2.783153	4657	0.2783153	0.559
7	2.5747	5775	0.25747	0.693
8	2.10549	6241	0.210549	0.749
9	1.975212	6994	0.1975212	0.839
10	1.872166	7841	0.1872166	0.941
15	1.624065	12236	0.1624065	1.468
21	1.89	16702	0.189	2.004
24	1.702	18623	0.1702	2.235



EXP NO: 09 DATE:21/02/24	MAC Protocols

SLOTTED ALOHA:

AIM:

To study and analyze slotted ALOHA system and plot the characteristic curve of throughput versus offer traffic for the same.

SOFTWARE REQUIRED:

NETSIM

THEORY:

ALOHA provides a wireless data network. It is a multiple access protocol (this protocol is for allocating a multiple access channel). There are two main versions of ALOHA: pure and slotted. They differ with respect to whether or not time is divided up into discrete slots into which all frames must fit.

In slotted ALOHA, time is divided up into discrete intervals, each interval corresponding to one frame. In Slotted ALOHA, a computer is required to wait for the beginning of the next slot in order to send the next packet. The probability of no other traffic being initiated during the entire vulnerable period is given by $e{-}2G$ which leads to $S= G\times e{-}2G$

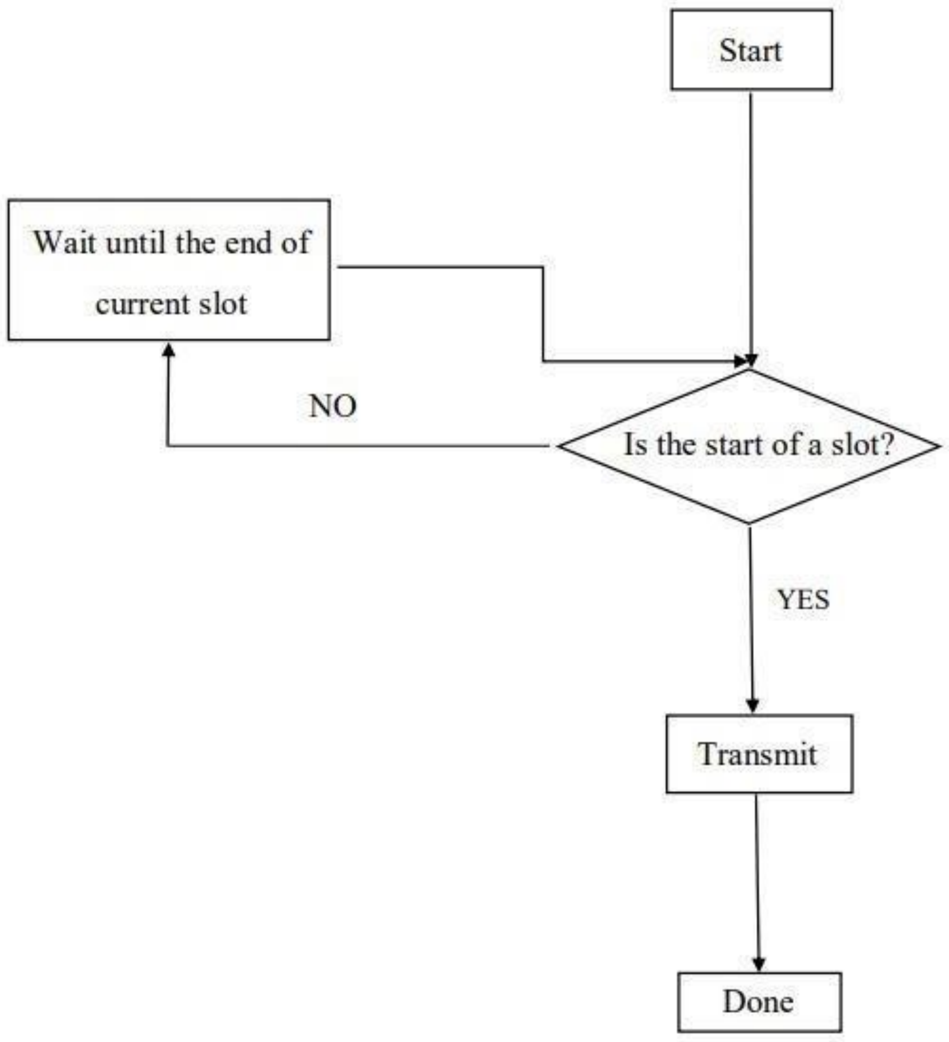
S ➔ the mean of the Poisson distribution with which frames are being generated. (frames per frame time)
G ➔ the mean of the Poisson distribution followed by the transmission attempts per frame time.
For reasonable throughput S should lie between 0 and 1.

FORMULA:

Attempts per packet time (G) can be calculated as follows;

$$G = \frac{\text{No. of packets transmitted x PT}}{\text{ST x 1000}}$$

FLOWCHART


Start
Wait until the end of current slot
NO
Is the start of a slot?
YES
Transmit
Done

G ➔ Attempts per packet time
PT ➔ Packet time (in seconds)
ST ➔ Simulation time (in seconds)

The throughput (in Mbps) per packet time can be obtained as follows:

$$S = \frac{\text{Throughput}(\text{in Mbps}) * 1000 * \text{PT}}{\text{PS} * 8}$$

S ➔ Throughput per packet time
PT ➔ Packet time (in milliseconds)
PS ➔ Packet size (in bytes)

Packet size (PS) =1472 (Data Size) + 28 (Overheads) = 1500 bytes
Packet time = 1.8 millisec (Bandwidth=10Mbps)

PROCEDURE:

- For creating scenarios, we select a new file for slotted ALOHA using legacy network.
- The requires components for slotted ALOHA is available. Initially 2 nodes are dropped and application is created.
- Edit the wireless node properties for generating traffic.

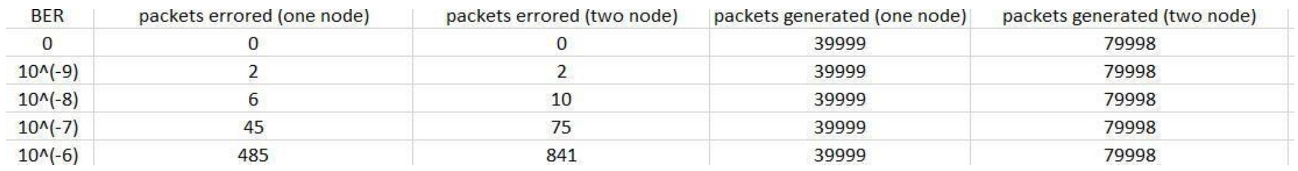
TCP ➔ Disable
Slot length ➔ 1500
Data rate ➔ 10Mbps

- Edit the application icon properties as

Application method	➔	Unicast
Application type	➔	Custom
Source ID	➔	1
Destination ID	➔	2
Packet size	➔	Exponential 1472
Inter arrival time	➔	Exponential 20000 μs

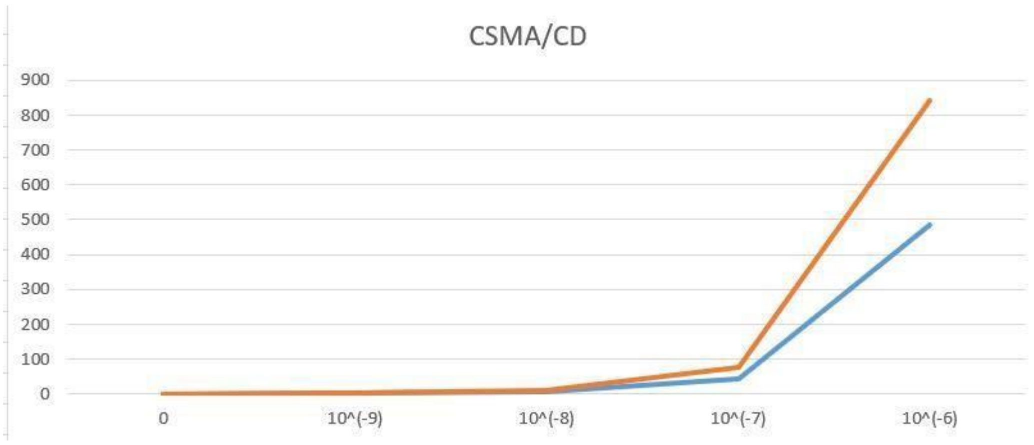
- Set the simulation time as 10s.
- Now repeat the process for different no. of nodes 3, 4, 6, 8 so on.

CSMA/CD:



BER	packets errored (one node)	packets errored (two node)	packets generated (one node)	packets generated (two node)
0	0	0	39999	79998
10^(-9)	2	2	39999	79998
10^(-8)	6	10	39999	79998
10^(-7)	45	75	39999	79998
10^(-6)	485	841	39999	79998

Transmitted packet time vs Throughput per packet time



➢ Note and tabulate the n values for various nodes.

INFERENCE:

From the graph of No. of packets, transmitted packet time vs throughput per packet time, we can infer that there if a steady increase in graph and later it decreases. When average no. of packets per slot increases, the throughput per slot also increases. Conversely since there is a high probability of collision as the no. of nodes N increases, the throughput per slot decreases.

CSMA/CD:

AIM:

To understand the impact of bit error rate on packet error and investigate the impact of error on a simple hub based CSMA / CD network.

SOFTWARE REQUIRED:

NETSIM

THEORY:

Bit error rate (BER): The bit error rate or bit error ratio is the number of bit errors divided by the total number of transferred bits during a studied time interval i.e.

$$\text{BER} = \frac{\text{Bit errors}}{\text{Total number of bits}}$$

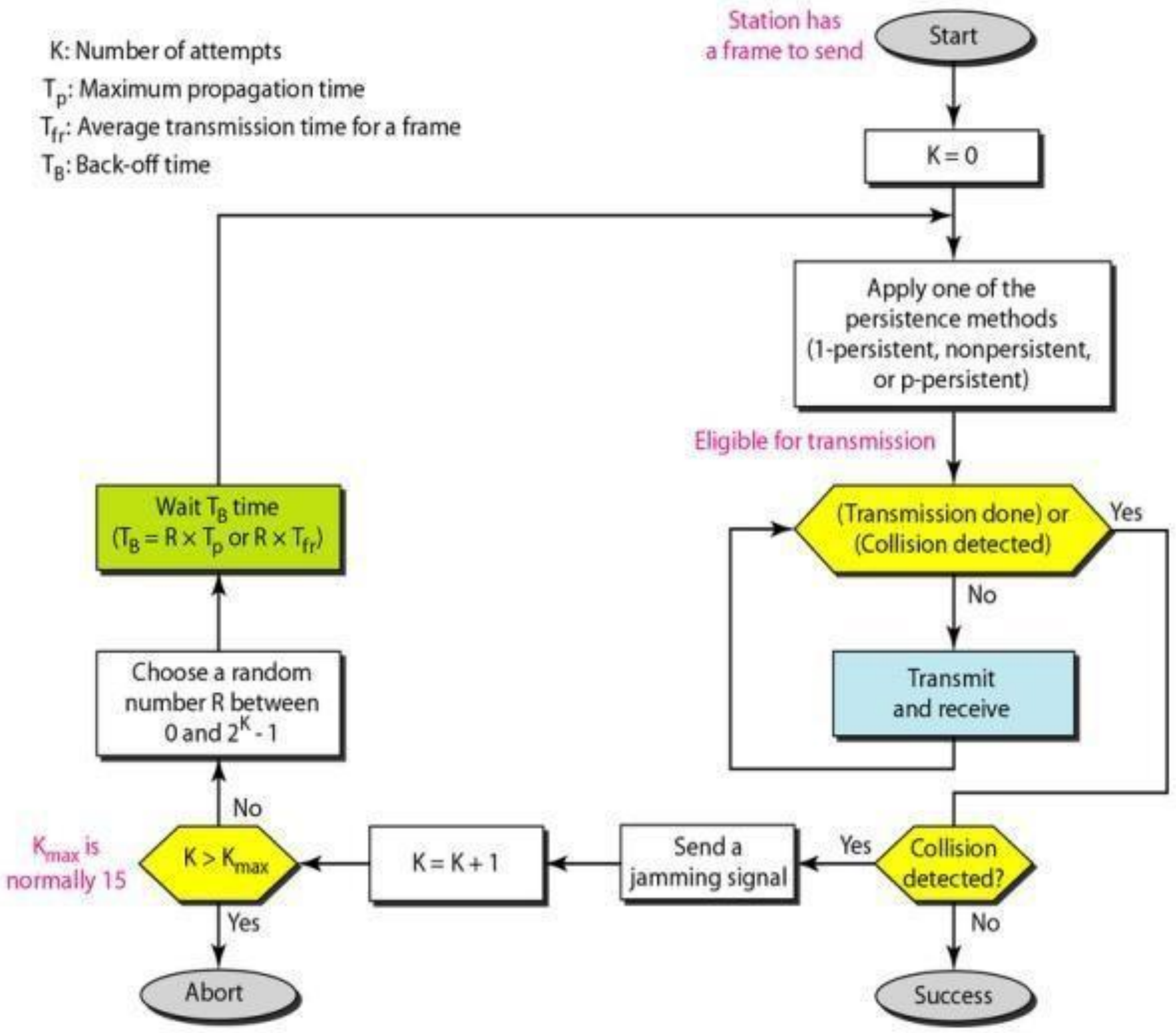
As BER coupler, the effect of all the components in the system given the actual performances of the system that has to be tested. Bit error probability is the expectation value of the BER. The BER can be considered as an approximate estimate of the bit error probability. This estimate is accurate for a long time interval and a high number of bit errors.

A packet is incorrectly received even if one bit is errorless

$$P_p = 1 - [1 - P_e]^N$$

P_p ➔ Packet error probability
P_e ➔ Bit error probability
N ➔ length of N bits in a packet.

FLOWCHART



K: Number of attempts
T_p: Maximum propagation time
T_{fr}: Average transmission time for a frame
T_B: Back-off time
Station has a frame to send
Start
K = 0
Apply one of the persistence methods (1-persistent, nonpersistent, or p-persistent)
Eligible for transmission
(Transmission done) or (Collision detected)
Yes
No
Transmit and receive
Collision detected?
Yes
No
Success
Send a jamming signal
K = K + 1
$K > K_{max}$
K_{max} is normally 15
Yes
Abort
No
Choose a random number R between 0 and $2^K - 1$
Wait T_B time ($T_B = R \times T_p$ or $R \times T_{fr}$)

PROCEDURE:

- For creating a new scenario we select legacy networks and then CSMA/CD.
- From the available components in the network, drag and drop one hub and two nodes, Node 1 and 2.
- Connect them using wires. Their properties are:

Link properties	Sample 1	Sample 2	Sample 3	Sample 4	Sample 5
Data rate(Mbps)	10	10	10	10	10
Bit error rate (BER)	No error	10^{-9}	10^{-8}	10^{-7}	10^{-6}

And set of application

Application method	➔	Unicast
Application type	➔	Custom
Source ID	➔	1
Destination ID	➔	2
Packet size	➔	Constant 1472
Packet interval time	➔	Constant 2500 μs

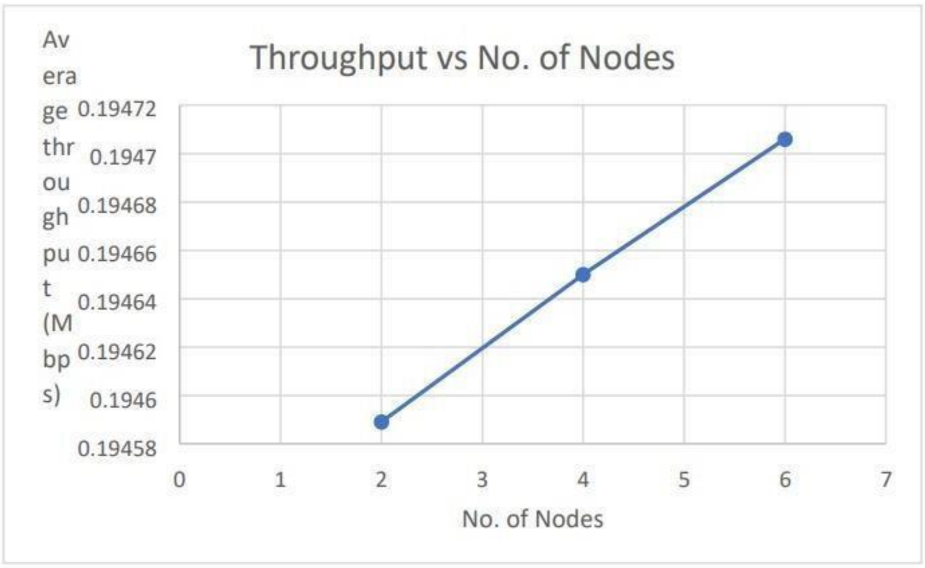
- Run the simulation and note down the values.
- For the next step, click and drop one bit with 4 nodes and connect them using wires.
- Set the same properties as above and note down the table for packet generated.
- Plot the graph for packets errored and bit error rate for both node transmission.

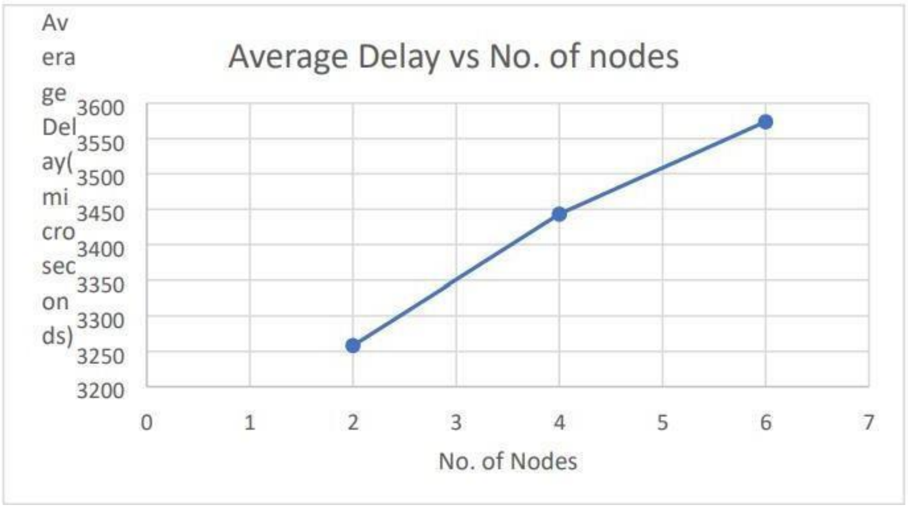
I NFERENCE:

From the graph of CSMA/CD, we get that when BER increases, packets errored also increases. This is because when there is an error in the bits, chance of packets begin correctly decreases.

The performance of CSMA/CD network and minimum bit error rate that can be tolerated without causing excessive packet error is calculated using graph.

CSMA/CA:


Throughput vs No. of Nodes
Average throughput (Mbps)
0.19472
0.1947
0.19468
0.19466
0.19464
0.19462
0.1946
0.19458
0
1
2
3
4
5
6
7
No. of Nodes


Average Delay vs No. of nodes
Average Delay(microseconds)
3600
3550
3500
3450
3400
3350
3300
3250
3200
0
1
2
3
4
5
6
7
No. of Nodes

CSMA/CA:

AIM:

To analyse the performance of a MANET running CSMA/CA in MAC as the node density is increased and to plot
Throughput versus No. of nodes
Delay average versus No. of nodes

SOFTWARE REQUIRED:

NETSIM

THEORY:

Mobile Ad-Hoc Network (MANET) is a self-configuring network of mobile nodes connected by wireless links to form an arbitrary topology without the use of existing infrastructure.

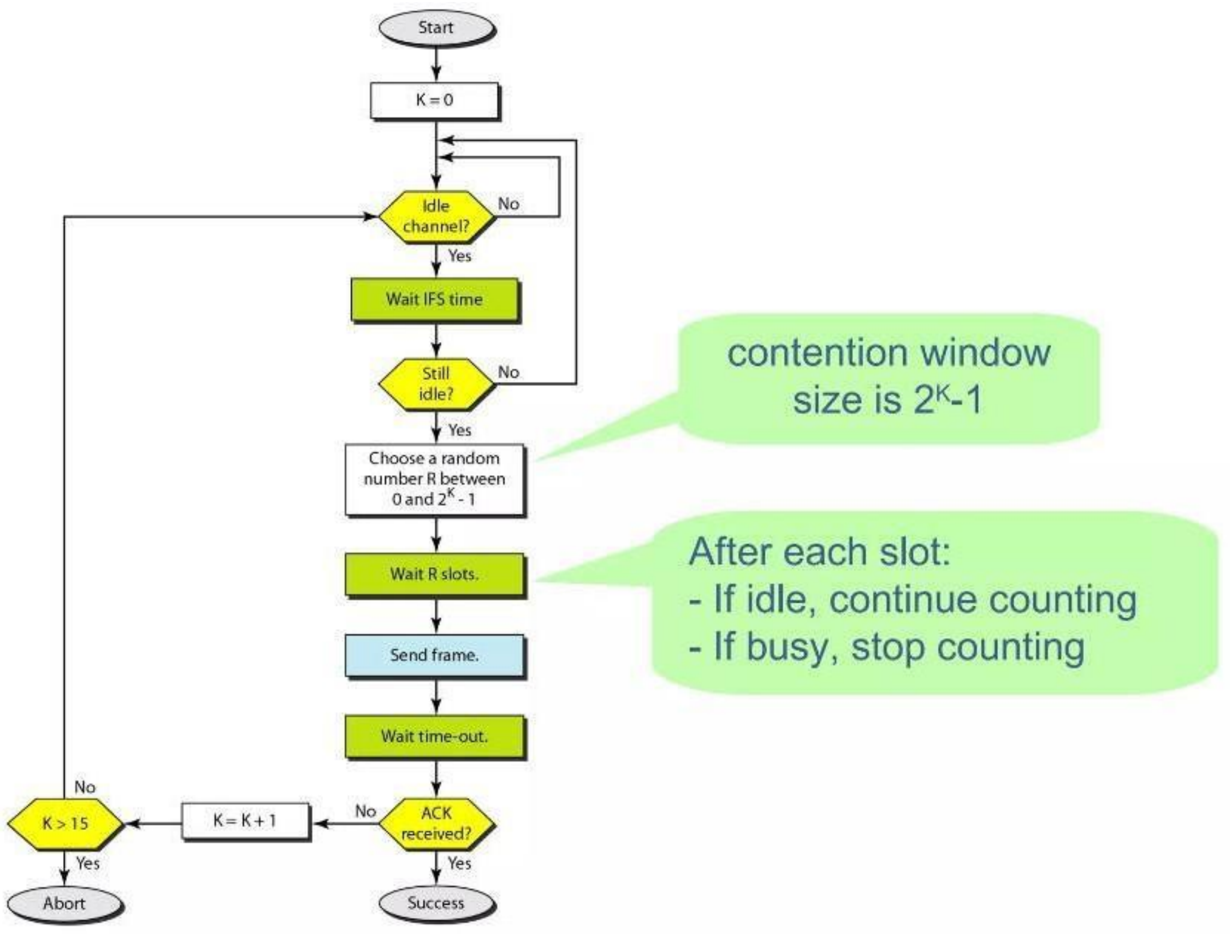
The nodes are free to move randomly. Thus the network's wireless topology may be unpredictable and may change rapidly.

The node density also has an impact on the routing performance. With very sparsely populated network the number of possible connection between any two nodes is very less and hence the performance is poor. It is expected that if the node density is increased the throughput of the network shall increase, but beyond a certain level if density is increased the performance degrades.

PROCEDURE:

- For creating new scenario, we select new legacy window network and then CSMA/CA.
- Then create the network using a wireless nodes. So from now, wireless nodes are increased.
- Correspondingly table below are 2 levels of simulation. Each level should be executed separately with the application properties mentioned.

FLOWCHART



Start
K = 0
Idle channel?
No
Yes
Wait IFS time
Still idle?
No
Yes
Choose a random number R between 0 and 2^K - 1
Wait R slots.
Send frame.
Wait time-out.
ACK received?
No
Yes
K = K + 1
K > 15
No
Yes
Abort
Success
contention window size is 2^K-1
After each slot:
- If idle, continue counting
- If busy, stop counting

Level	**Wireless Node**	**X Ordinate**	**Y Ordinate**
1	1	50	100
	2	100	150
2	1	50	100
	2	100	150
	3	75	75
	4	125	125
3	1	50	100
	2	100	150
	3	80	70
	4	140	130
	5	110	40
	6	160	90

- Disable TCP in all wireless nodes.
- Right click on grid area to modify wireless node properties.

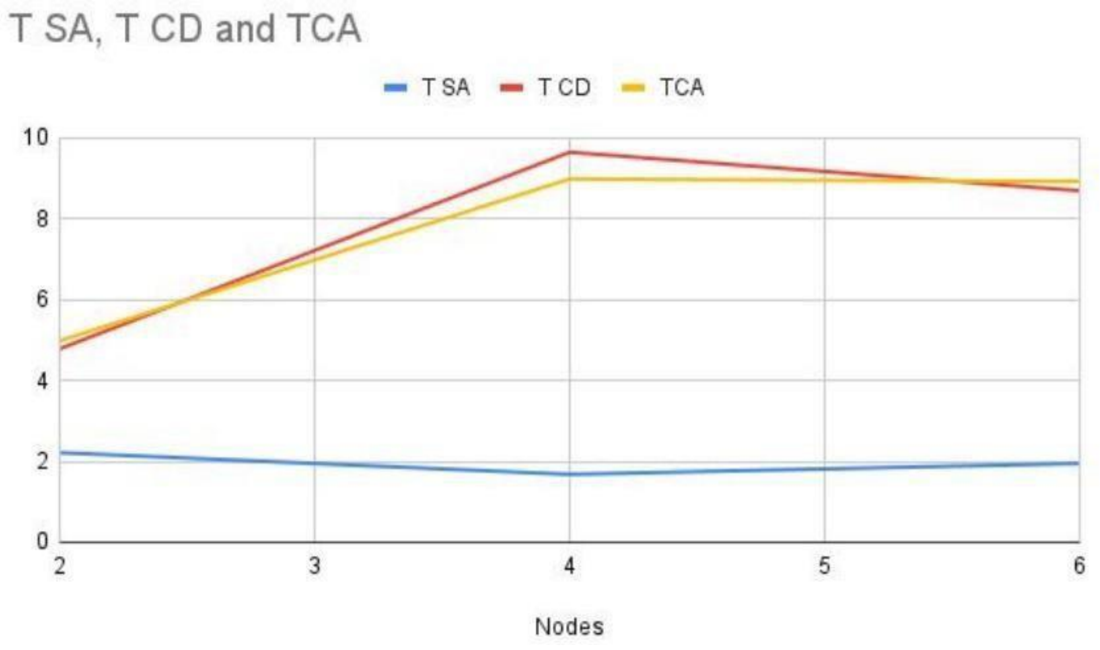
Path loss model	➔	log distance
Path loss component	➔	2

- Then we have to model the traffic in network using application after selecting the applications dialog box. Set the below mentioned values:

Application type	➔	Custom
Source ID	➔	As per levels source ID is varied as per nodes
Destination ID	➔	Same rule applies for destination ID
Packet size		
Distribution	➔	Constant
Value (bytes)	➔	1472
Inter arrival time		
Distribution	➔	Constant
Value (μs)	➔	60000

- Save the schematic and note down the values. Repeat the same steps for level 2 and level 3with increasing application.

COMPARISON GRAPH:



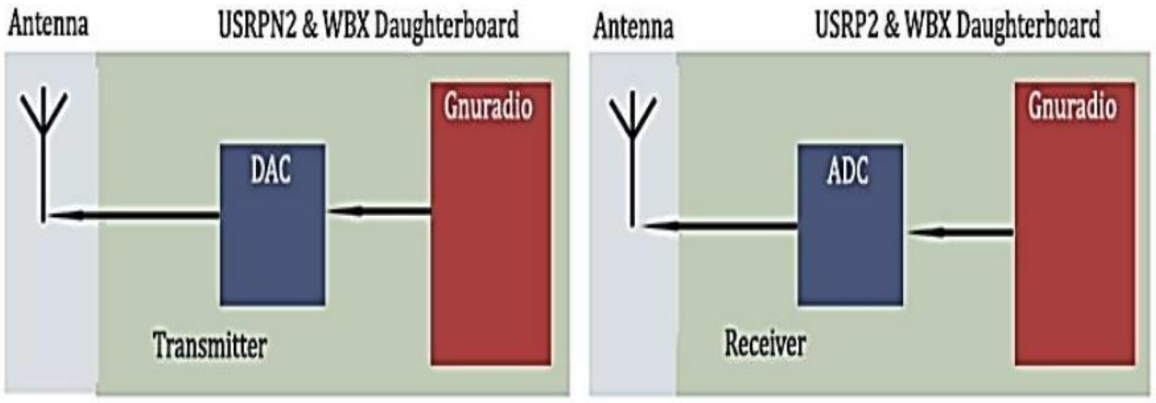
INFERENCE:

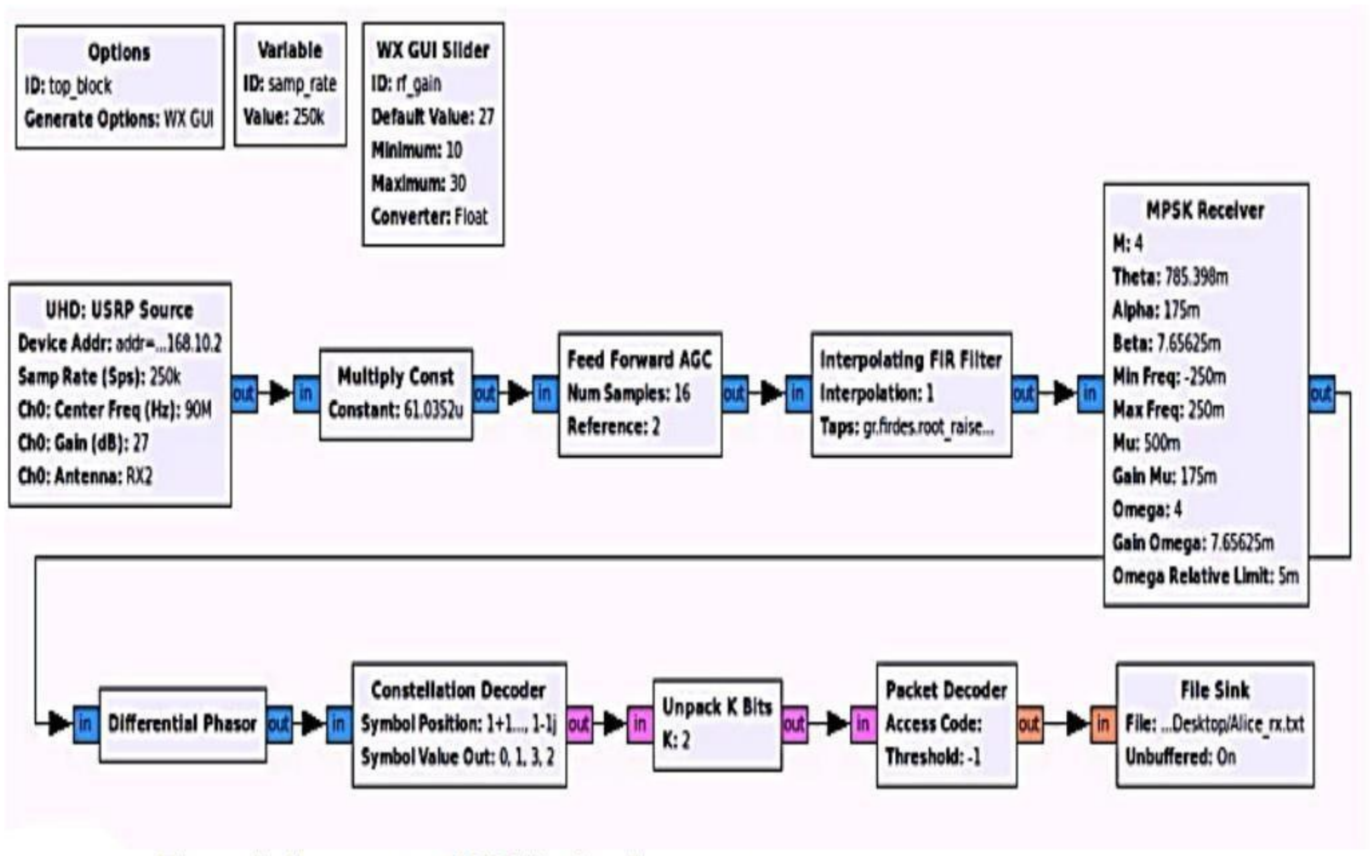
From the graph of average delay versus no. of nodes, when no. of nodes increases, average delay also increases. It takes longer for a packet to reach its destination.

From the graph of average throughput versus no. of nodes, the average throughput increases for well designed network.

RESULT:

MAC protocol namely slotted ALOHA, CSMA/CA, CSMA/CD are implemented and verified using NETSIM.





Options
ID: top_block
Generate Options: WX GUI

Variable
ID: samp_rate
Value: 250k

WX GUI Slider
ID: rf_gain
Default Value: 27
Minimum: 10
Maximum: 30
Converter: Float

UHD: USRP Source
Device Addr: addr=...168.10.2
Samp Rate (Sps): 250k
Ch0: Center Freq (Hz): 90M
Ch0: Gain (dB): 27
Ch0: Antenna: RX2

Multiply Const
Constant: 61.0352u

Feed Forward AGC
Num Samples: 16
Reference: 2

Interpolating FIR Filter
Interpolation: 1
Taps: gr.firdes.root_raise...

MPSK Receiver
M: 4
Theta: 785.398m
Alpha: 175m
Beta: 7.65625m
Min Freq: -250m
Max Freq: 250m
Mu: 500m
Gain Mu: 175m
Omega: 4
Gain Omega: 7.65625m
Omega Relative Limit: 5m

Differential Phasor

Constellation Decoder
Symbol Position: 1+1..., 1-1j
Symbol Value Out: 0, 1, 3, 2

Unpack K Bits
K: 2

Packet Decoder
Access Code:
Threshold: -1

File Sink
File: ...Desktop/Alice_rx.txt
Unbuffered: On

Expanded system in GNU Radio Companion

EXP NO: 10 DATE:20/03/24	**Study of SDR based transceiver using USRP and GNU radio**

AIM:

To study about software defined ratio: SDR based transceiver of digital communication system using universal software radio peripheral: USRP family of products.

To simulate OFDM and to study its characteristics.

SOFTWARE REQUIRED:

GNU Radio

HARDWARE REQUIRED:

USRP kit

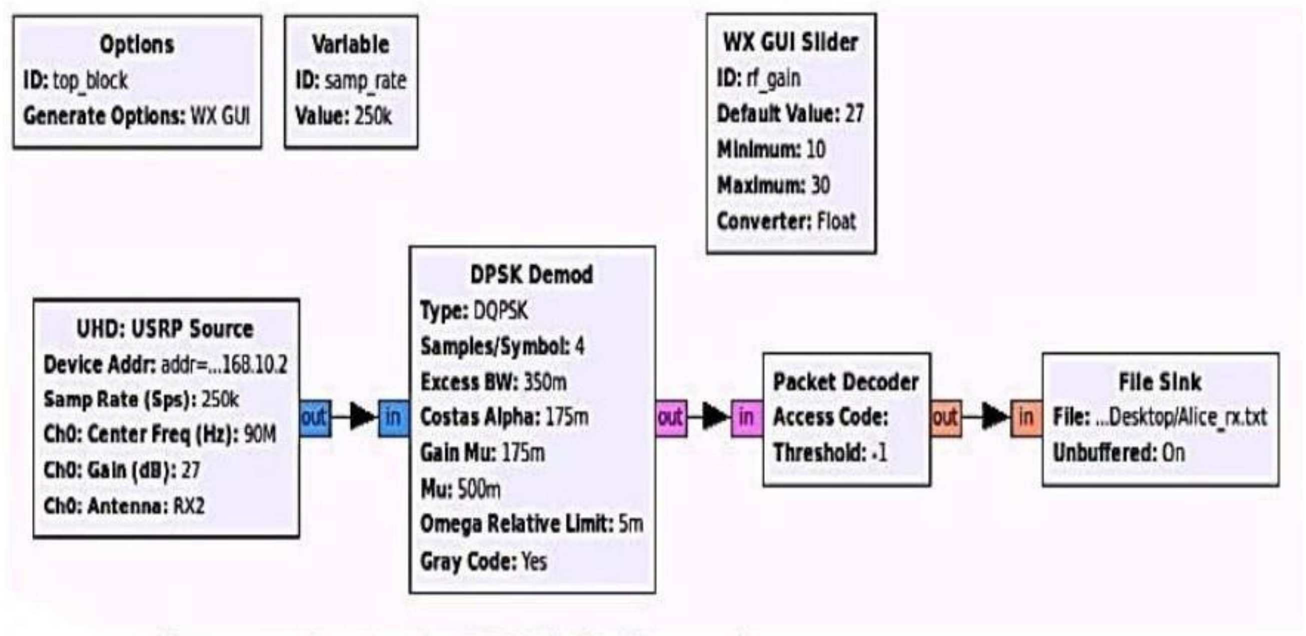
THEORY:

GNU Radio:

GNU Radio is a framework that enables users to design, simulate, and deploy highly capable real-world radio systems. It is a highly modular, "flowgraph"-oriented framework that comes with a comprehensive library of processing blocks that can be readily combined to make complex signal processing applications. GNU Radio has been used for a huge array of real-world radio applications, including audio processing, mobile communications, tracking satellites, radar systems, GSM networks, Digital Radio Mondiale, and much more
- all in computer software. It is, by itself, not a solution to talk to any specific hardware. Nor does it provide outof the-box applications for specific radio communications standards (e.g., 802.11, ZigBee,LTE, etc.,), but it can be (and has been) used to develop implementations of basically any band-limited communication standard. GNU Radio is a framework dedicated to writing signal processing applications for commodity computers. GNU Radio wraps functionality in easy-to-use reusable blocks, offers excellent scalability, provides an extensive library of standard algorithms, and is heavily optimized for a large variety of common platforms.

USRP kit:

USRP Software Defined Radio device provides a software defined RF architecture to design, prototype and deploy wireless systems with custom signal processing. USRP Software Defined Radio Devices also include options with an onboard FPGA—an SDR game changer that provides wireless communications designers an affordable SDR with unprecedented performance for developing



Options
ID: top_block
Generate Options: WX GUI

Variable
ID: samp_rate
Value: 250k

WX GUI Slider
ID: rf_gain
Default Value: 27
Minimum: 10
Maximum: 30
Converter: Float

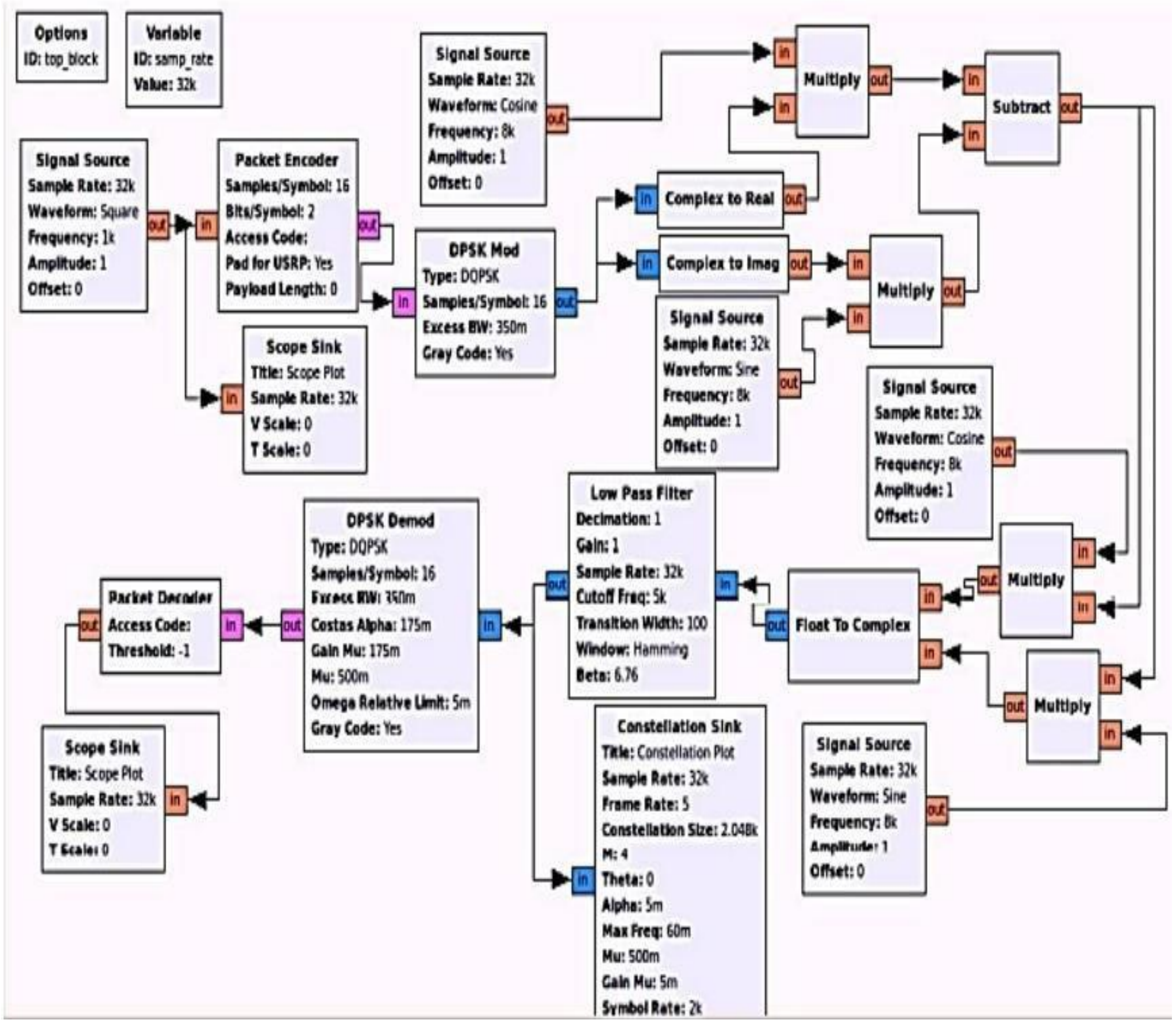
UHD: USRP Source
Device Addr: addr=...168.10.2
Samp Rate (Sps): 250k
Ch0: Center Freq (Hz): 90M
Ch0: Gain (dB): 27
Ch0: Antenna: RX2

DPSK Demod
Type: DQPSK
Samples/Symbol: 4
Excess BW: 350m
Costas Alpha: 175m
Gain Mu: 175m
Mu: 500m
Omega Relative Limit: 5m
Gray Code: Yes

Packet Decoder
Access Code:
Threshold: -1

File Sink
File: ...Desktop/Alice_rx.txt
Unbuffered: On

Compressed system in GNU Radio Companion



Options
ID: top_block

Variable
ID: samp_rate
Value: 32k

Signal Source
Sample Rate: 32k
Waveform: Cosine
Frequency: 8k
Amplitude: 1
Offset: 0

Signal Source
Sample Rate: 32k
Waveform: Square
Frequency: 1k
Amplitude: 1
Offset: 0

Packet Encoder
Samples/Symbol: 16
Bits/Symbol: 2
Access Code:
Pad for USRP: Yes
Payload Length: 0

DPSK Mod
Type: DQPSK
Samples/Symbol: 16
Excess BW: 350m
Gray Code: Yes

Scope Sink
Title: Scope Plot
Sample Rate: 32k
V Scale: 0
T Scale: 0

Signal Source
Sample Rate: 32k
Waveform: Sine
Frequency: 8k
Amplitude: 1
Offset: 0

Signal Source
Sample Rate: 32k
Waveform: Cosine
Frequency: 8k
Amplitude: 1
Offset: 0

Low Pass Filter
Decimation: 1
Gain: 1
Sample Rate: 32k
Cutoff Freq: 5k
Transition Width: 100
Window: Hamming
Beta: 6.76

DPSK Demod
Type: DQPSK
Samples/Symbol: 16
Excess BW: 350m
Costas Alpha: 175m
Gain Mu: 175m
Mu: 500m
Omega Relative Limit: 5m
Gray Code: Yes

Packet Decoder
Access Code:
Threshold: -1

Scope Sink
Title: Scope Plot
Sample Rate: 32k
V Scale: 0
T Scale: 0

Constellation Sink
Title: Constellation Plot
Sample Rate: 32k
Frame Rate: 5
Constellation Size: 2.048k
M: 4
Theta: 0
Alpha: 5m
Max Freq: 60m
Mu: 500m
Gain Mu: 5m
Symbol Rate: 2k

Signal Source
Sample Rate: 32k
Waveform: Sine
Frequency: 8k
Amplitude: 1
Offset: 0

next generation 5G wireless communication systems. USRP Software Defined Radio Devices have an optional onboard GPS that you can use to synchronize multiple software defined radios and enable advanced application possibilities such as distributed multi-channel synchronized systems.

The USRP Software Defined Radio Device is a reconfigurable RF device that includes a combination of host based processors, FPGAs, and RF front ends. The USRP Software Defined Radio Device include options that range from lower cost options with fixed FPGA personalities to high end radios with a large, open FPGAs and wide instantaneous bandwidth. These devices can be used for applications such as multiple input, multiple output (MIMO) and LTE/WiFi testbeds, SIGINT, and radar systems.

OFDM:

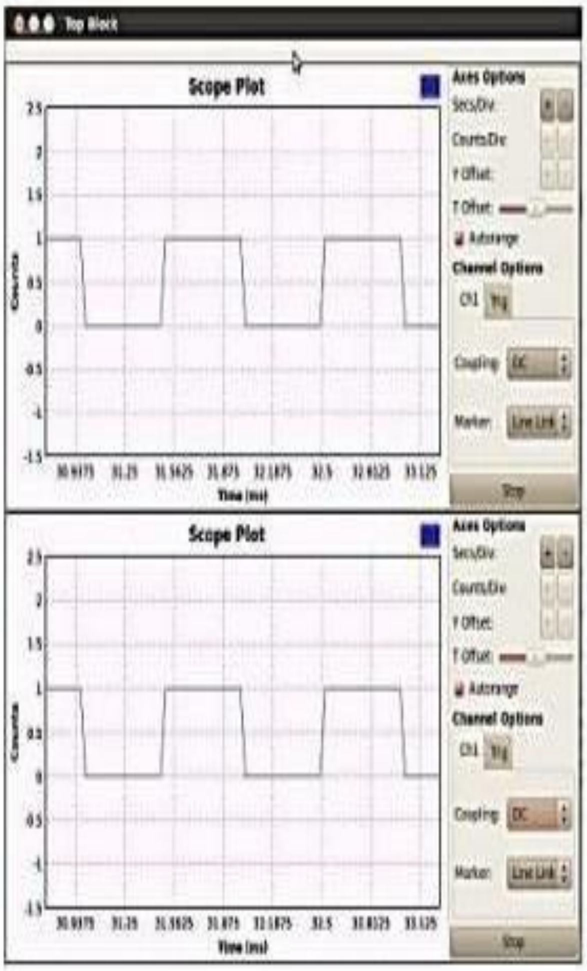
- Orthogonal Frequency Division Multiplexing (OFDM) is a digital multi-carrier modulation scheme that extends the concept of single subcarrier modulation by using multiple subcarriers within the same single channel. Rather than transmit a high-rate stream of data with a single subcarrier, OFDM makes use of a large number of closely spaced orthogonal subcarriers that are transmitted in parallel.

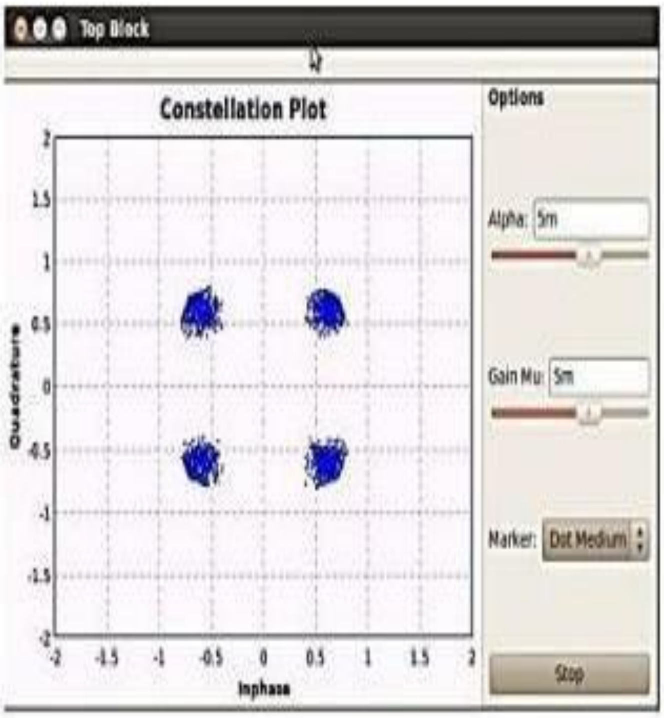
- Each subcarrier is modulated with a conventional digital modulation scheme (such as QPSK, 16QAM, etc.) at low symbol rate. However, the combination of many subcarriers enables data rates similar to conventional single-carrier modulation schemes within equivalent bandwidths.

- The OFDM signal can be described as a set of closely spaced FDM subcarriers. In the frequency domain, each transmitted subcarrier results in a sinc function spectrum with side lobes that produce overlapping spectra between subcarriers. This results in subcarrier interference except at orthogonally spaced frequencies.

- At orthogonal frequencies, the individual peaks of subcarriers all line up with the nulls of the other subcarriers. This overlap of spectral energy does not interfere with the system's ability to recover the original signal. The receiver multiplies (i.e., correlates) the incoming signal by the known set of sinusoids to recover the original set of bits sent.

- The use of orthogonal subcarriers allows more subcarriers per bandwidth resulting in an increase in spectral efficiency. In a perfect OFDM signal, Orthogonality prevents interference between overlapping carriers.



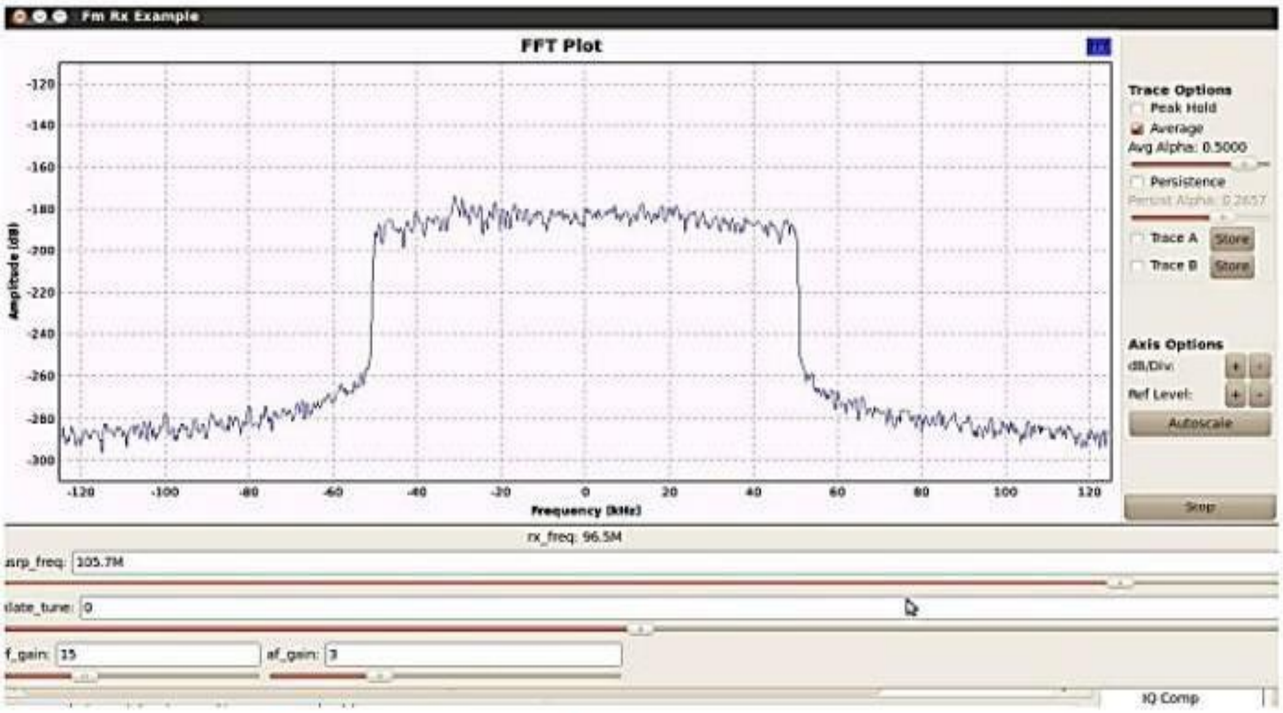


Alpha: 5m

Gain Mu: 5m

Marker: Dot Medium

Left: Transmitted signal and Received signal. Right: The constellation plot of receiver



Avg Alpha: 0.5000

Persist Alpha: 0.2657

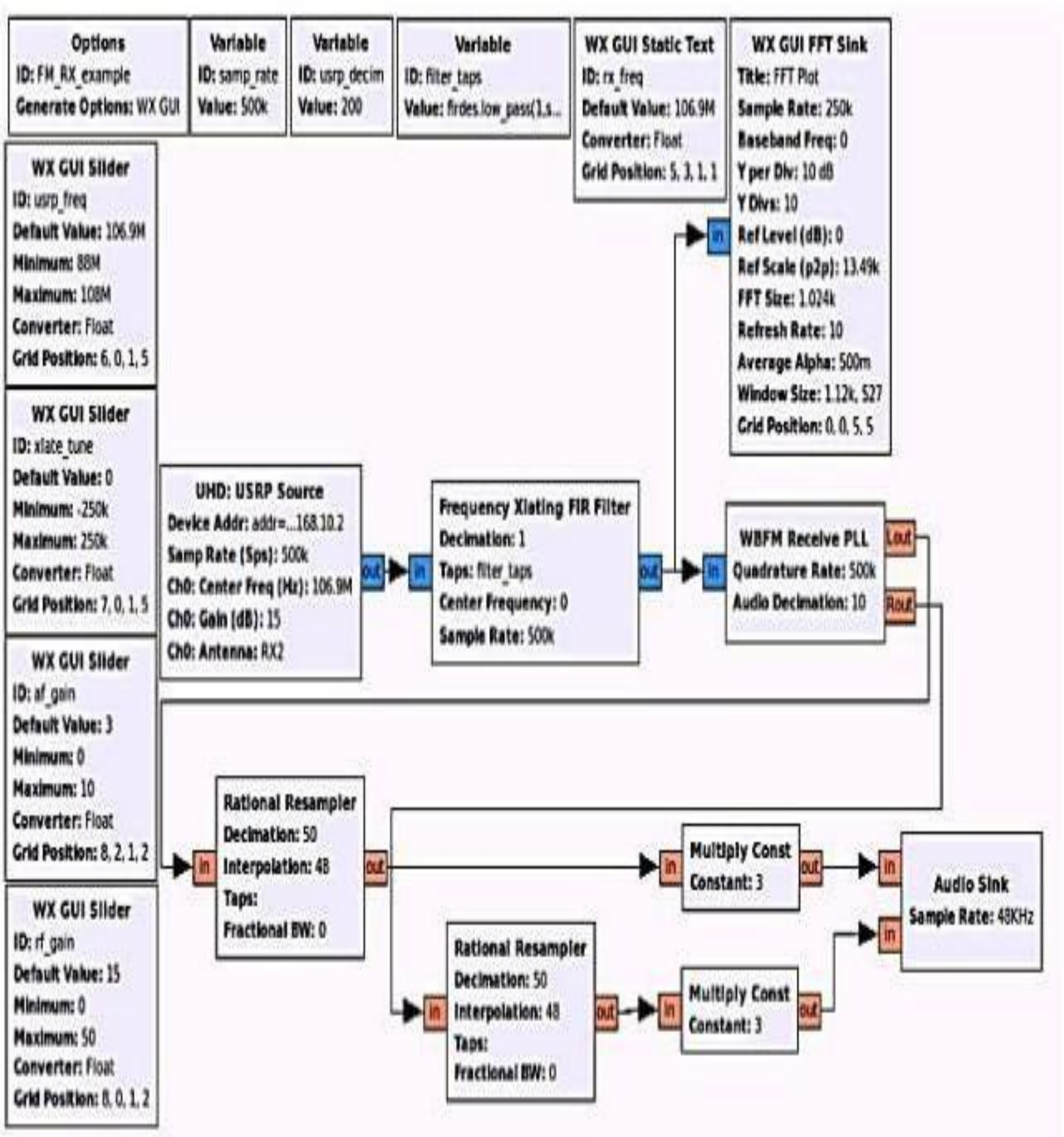
rx_freq: 96.5M

usrp_freq: 105.7M

xlate_tune: 0

rf_gain: 15 af_gain: 3

Received Spectrum of FM Receiver



Options
ID: FM_RX_example
Generate Options: WX GUI

Variable
ID: samp_rate
Value: 500k

Variable
ID: usrp_decim
Value: 200

Variable
ID: filter_taps
Value: firdes.low_pass(1,s...

WX GUI Static Text
ID: rx_freq
Default Value: 106.9M
Converter: Float
Grid Position: 5, 3, 1, 1

WX GUI FFT Sink
Title: FFT Plot
Sample Rate: 250k
Baseband Freq: 0
Y per Div: 10 dB
Y Divs: 10
Ref Level (dB): 0
Ref Scale (p2p): 13.49k
FFT Size: 1.024k
Refresh Rate: 10
Average Alpha: 500m
Window Size: 1.12k, 527
Grid Position: 0, 0, 5, 5

WX GUI Slider
ID: usrp_freq
Default Value: 106.9M
Minimum: 88M
Maximum: 108M
Converter: Float
Grid Position: 6, 0, 1, 5

WX GUI Slider
ID: xlate_tune
Default Value: 0
Minimum: -250k
Maximum: 250k
Converter: Float
Grid Position: 7, 0, 1, 5

WX GUI Slider
ID: af_gain
Default Value: 3
Minimum: 0
Maximum: 10
Converter: Float
Grid Position: 8, 2, 1, 2

WX GUI Slider
ID: rf_gain
Default Value: 15
Minimum: 0
Maximum: 50
Converter: Float
Grid Position: 8, 0, 1, 2

UHD: USRP Source
Device Addr: addr=...168.10.2
Samp Rate (Sps): 500k
Ch0: Center Freq (Hz): 106.9M
Ch0: Gain (dB): 15
Ch0: Antenna: RX2

Frequency Xlating FIR Filter
Decimation: 1
Taps: filter_taps
Center Frequency: 0
Sample Rate: 500k

WBFM Receive PLL
Quadrature Rate: 500k
Audio Decimation: 10

Rational Resampler
Decimation: 50
Interpolation: 48
Taps:
Fractional BW: 0

Rational Resampler
Decimation: 50
Interpolation: 48
Taps:
Fractional BW: 0

Multiply Const
Constant: 3

Multiply Const
Constant: 3

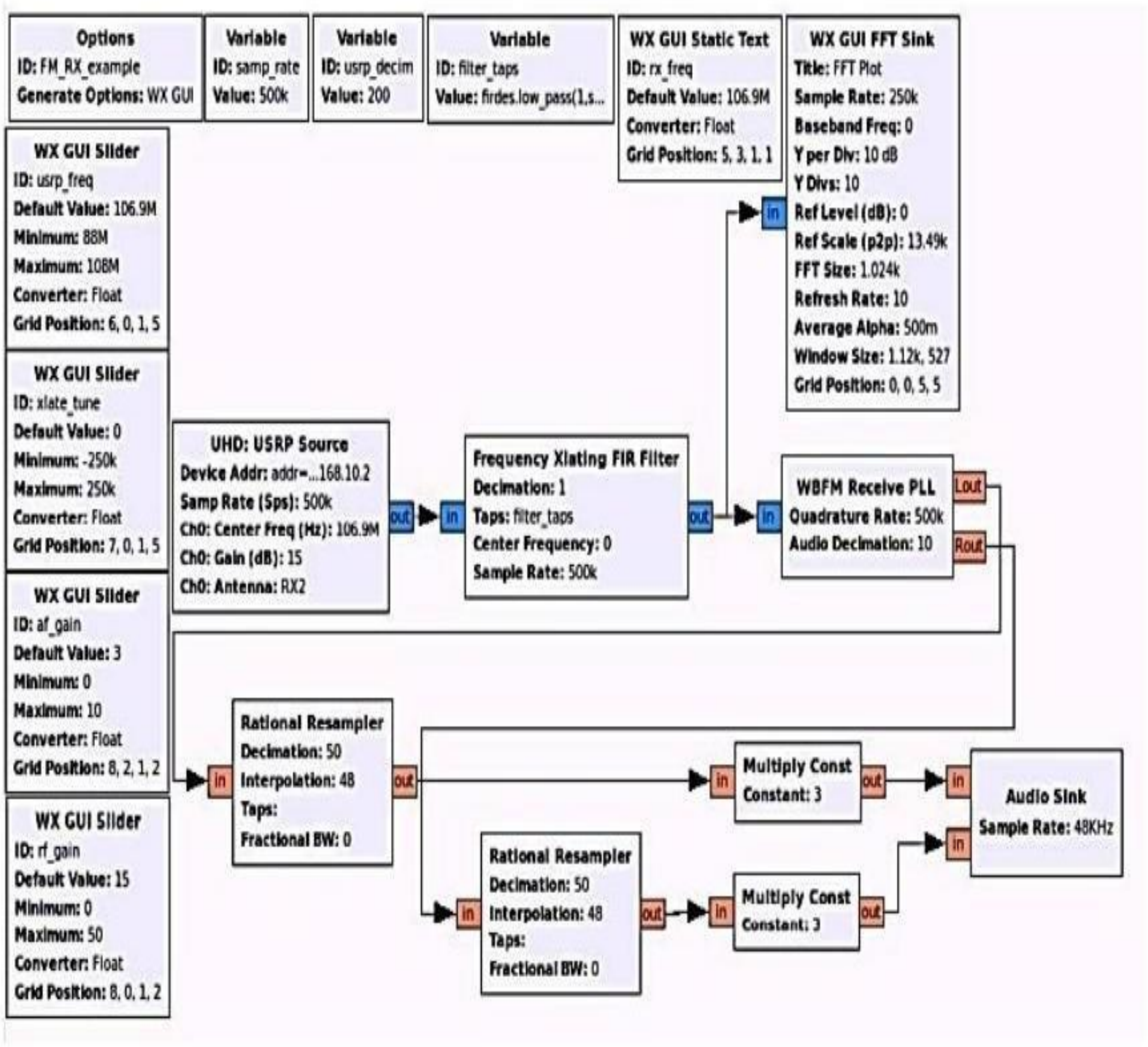
Audio Sink
Sample Rate: 48KHz

FM Radio Rx System

PROCEDURE:

GNU Radio Companion (GRC) is a graphical user interface that allows you to build GNU radio flow graphs.

1. Open a terminal window using: Applications > Accessories > Terminal. At the prompt type: grc
2. An untitled GRC window should open.
3. Double click on the Options block. This block sets some general parameters for the flow graph. Leave the ID as top_block. Type in a project title (such as Tutorial 1) and author. Set Generate Options to WX GUI, Run to Autostart, and Realtime Scheduling to Off. Then close the properties window. The other block that is present is the Variable block. It is used to set the sample rate.
4. On the right side of the window is a list of the blocks that are available. By expanding any of the categories (click on the triangle to the left) you can see the blocks available.
5. Open the Sources category and double click on the Signal Source. Note that a signal source block will now appear in the main window. Double click on the block and the properties window will open. In order to view this wave, we need one of the graphical sinks. Expand the Graphical Sink category and double click on the Scope Sink. It should appear in the main window. Double click on the block and change the Type to Float. Leave the other Parameters at their default values and close the properties window.
6. In order to observe the operation of this simple system we must generate the flow graph and then execute it. Click first on the "Generate the flow graph" icon. Click next on the "Execute the flow graph" icon to view the graph.
7. Open a file browser in Ubuntu (Places > Home Folder). Go to the directory that contains the GRC file that you have been working on. If you are unsure as to where this is, the path to this file is shown in the bottom portion of the GRC window. In addition to saving a ".grc" file with your flow graph, note that there is also a file titled "top_block.py". Double click on this block. You will be given the option to Run or Display this file. Select Display. This is the Python file that is generated by GRC. It is the file that is being run when you execute the flow graph. You can modify this file and run it from the terminal window. This allows you to use features that are not included in GRC. Keep in mind that every time you run your flow graph in GRC, it will overwrite the Python script that is generated. So, if you make changes directly in the Python Script that you want to keep, save it under another name.


Options
ID: FM_RX_example
Generate Options: WX GUI

Variable
ID: samp_rate
Value: 500k

Variable
ID: usrp_decim
Value: 200

Variable
ID: filter_taps
Value: firdes.low_pass(1,s...

WX GUI Static Text
ID: rx_freq
Default Value: 106.9M
Converter: Float
Grid Position: 5, 3, 1, 1

WX GUI FFT Sink
Title: FFT Plot
Sample Rate: 250k
Baseband Freq: 0
Y per Div: 10 dB
Y Divs: 10
Ref Level (dB): 0
Ref Scale (p2p): 13.49k
FFT Size: 1.024k
Refresh Rate: 10
Average Alpha: 500m
Window Size: 1.12k, 527
Grid Position: 0, 0, 5, 5

WX GUI Slider
ID: usrp_freq
Default Value: 106.9M
Minimum: 88M
Maximum: 108M
Converter: Float
Grid Position: 6, 0, 1, 5

WX GUI Slider
ID: xlate_tune
Default Value: 0
Minimum: -250k
Maximum: 250k
Converter: Float
Grid Position: 7, 0, 1, 5

WX GUI Slider
ID: af_gain
Default Value: 3
Minimum: 0
Maximum: 10
Converter: Float
Grid Position: 8, 2, 1, 2

WX GUI Slider
ID: rf_gain
Default Value: 15
Minimum: 0
Maximum: 50
Converter: Float
Grid Position: 8, 0, 1, 2

UHD: USRP Source
Device Addr: addr=...168.10.2
Samp Rate (Sps): 500k
Ch0: Center Freq (Hz): 106.9M
Ch0: Gain (dB): 15
Ch0: Antenna: RX2

Frequency Xlating FIR Filter
Decimation: 1
Taps: filter_taps
Center Frequency: 0
Sample Rate: 500k

WBFM Receive PLL
Quadrature Rate: 500k
Audio Decimation: 10

Rational Resampler
Decimation: 50
Interpolation: 48
Taps:
Fractional BW: 0

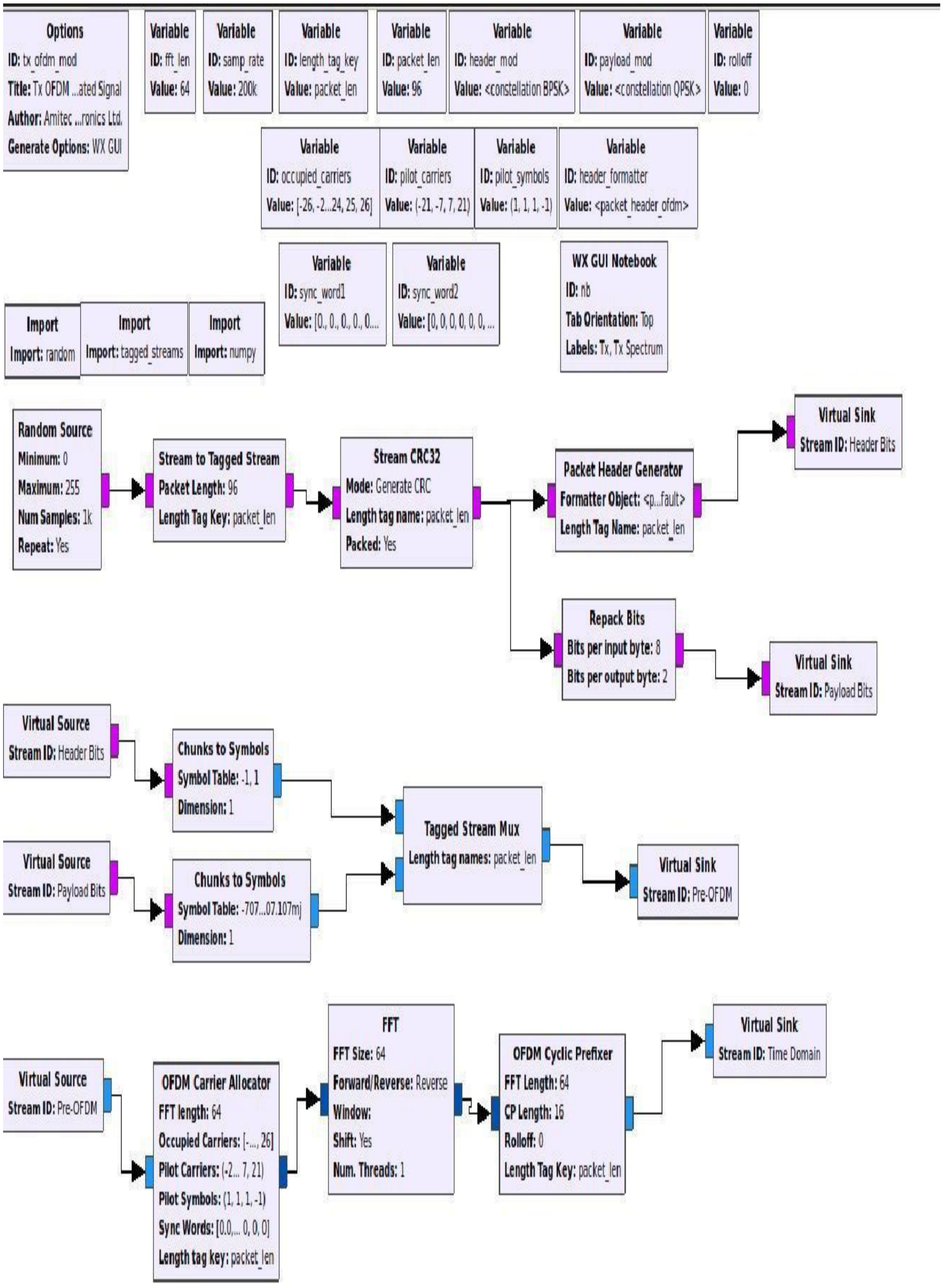
Rational Resampler
Decimation: 50
Interpolation: 48
Taps:
Fractional BW: 0

Multiply Const
Constant: 3

Multiply Const
Constant: 3

Audio Sink
Sample Rate: 48KHz

- *Complete flowgraph with appropriate connections, data types, and parameters*



Options
ID: tx_ofdm_mod
Title: Tx OFDM ...ated Signal
Author: Amitec ...ronics Ltd.
Generate Options: WX GUI

Variable
ID: fft_len
Value: 64

Variable
ID: samp_rate
Value: 200k

Variable
ID: length_tag_key
Value: packet_len

Variable
ID: packet_len
Value: 96

Variable
ID: header_mod
Value: <constellation BPSK>

Variable
ID: payload_mod
Value: <constellation QPSK>

Variable
ID: rolloff
Value: 0

Variable
ID: occupied_carriers
Value: [-26, -2...24, 25, 26]

Variable
ID: pilot_carriers
Value: (-21, -7, 7, 21)

Variable
ID: pilot_symbols
Value: (1, 1, 1, -1)

Variable
ID: header_formatter
Value: <packet_header_ofdm>

Variable
ID: sync_word1
Value: [0., 0., 0., 0., 0....

Variable
ID: sync_word2
Value: [0, 0, 0, 0, 0, 0, ...

WX GUI Notebook
ID: nb
Tab Orientation: Top
Labels: Tx, Tx Spectrum

Import
Import: random

Import
Import: tagged_streams

Import
Import: numpy

Random Source
Minimum: 0
Maximum: 255
Num Samples: 1k
Repeat: Yes

Stream to Tagged Stream
Packet Length: 96
Length Tag Key: packet_len

Stream CRC32
Mode: Generate CRC
Length tag name: packet_len
Packed: Yes

Packet Header Generator
Formatter Object: <p...fault>
Length Tag Name: packet_len

Virtual Sink
Stream ID: Header Bits

Repack Bits
Bits per input byte: 8
Bits per output byte: 2

Virtual Sink
Stream ID: Payload Bits

Virtual Source
Stream ID: Header Bits

Chunks to Symbols
Symbol Table: -1, 1
Dimension: 1

Virtual Source
Stream ID: Payload Bits

Chunks to Symbols
Symbol Table: -707...07.107mj
Dimension: 1

Tagged Stream Mux
Length tag names: packet_len

Virtual Sink
Stream ID: Pre-OFDM

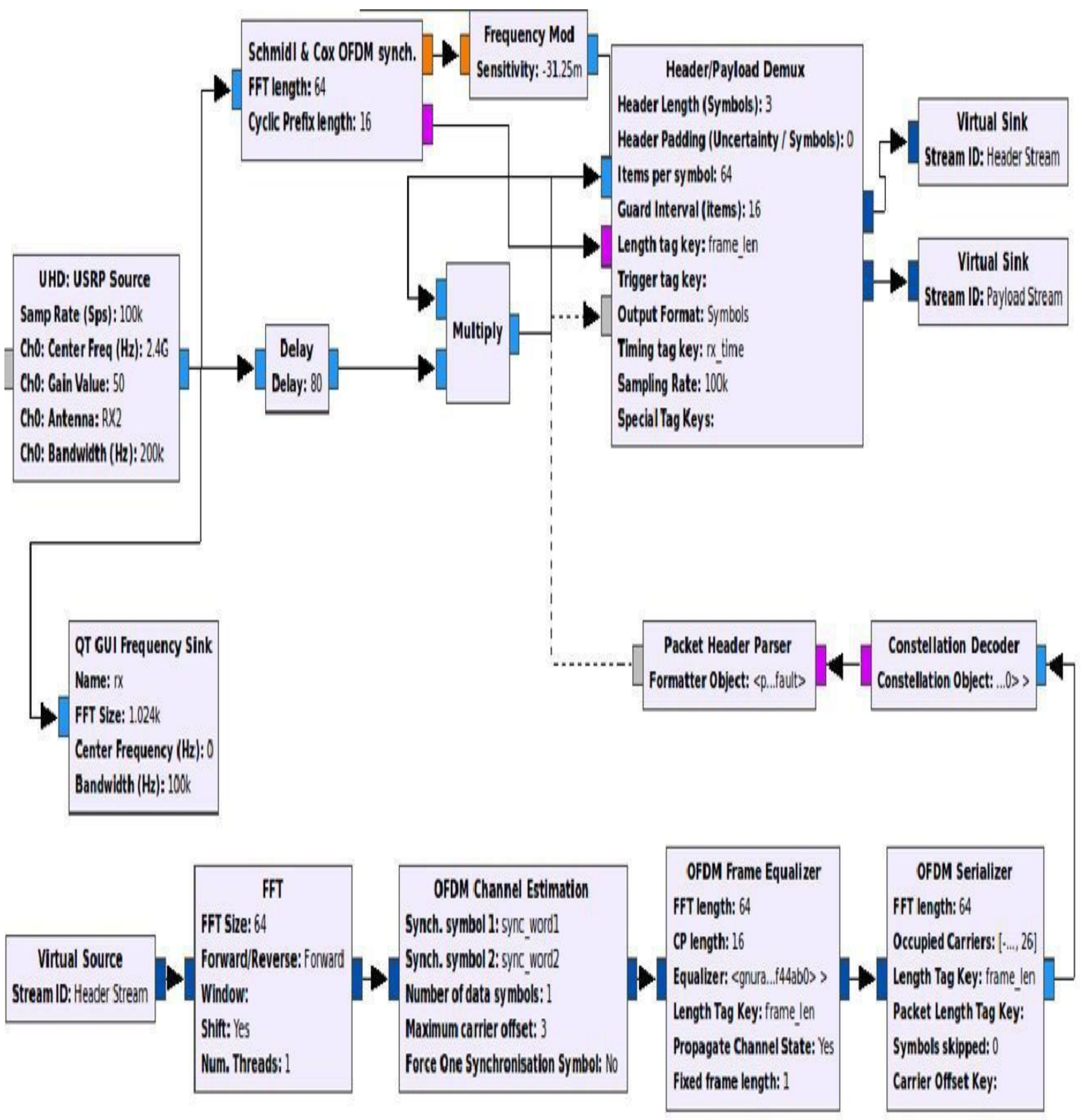
Virtual Source
Stream ID: Pre-OFDM

OFDM Carrier Allocator
FFT length: 64
Occupied Carriers: [-..., 26]
Pilot Carriers: (-2... 7, 21)
Pilot Symbols: (1, 1, 1, -1)
Sync Words: [0.0,... 0, 0, 0]
Length tag key: packet_len

FFT
FFT Size: 64
Forward/Reverse: Reverse
Window:
Shift: Yes
Num. Threads: 1

OFDM Cyclic Prefixer
FFT Length: 64
CP Length: 16
Rolloff: 0
Length Tag Key: packet_len

Virtual Sink
Stream ID: Time Domain


Schmidl & Cox OFDM synch.
FFT length: 64
Cyclic Prefix length: 16

Frequency Mod
Sensitivity: -31.25m

Header/Payload Demux
Header Length (Symbols): 3
Header Padding (Uncertainty / Symbols): 0
Items per symbol: 64
Guard Interval (items): 16
Length tag key: frame_len
Trigger tag key:
Output Format: Symbols
Timing tag key: rx_time
Sampling Rate: 100k
Special Tag Keys:

Virtual Sink
Stream ID: Header Stream

Virtual Sink
Stream ID: Payload Stream

UHD: USRP Source
Samp Rate (Sps): 100k
Ch0: Center Freq (Hz): 2.4G
Ch0: Gain Value: 50
Ch0: Antenna: RX2
Ch0: Bandwidth (Hz): 200k

Delay
Delay: 80

Multiply

QT GUI Frequency Sink
Name: rx
FFT Size: 1.024k
Center Frequency (Hz): 0
Bandwidth (Hz): 100k

Packet Header Parser
Formatter Object: <p...fault>

Constellation Decoder
Constellation Object: ...0> >

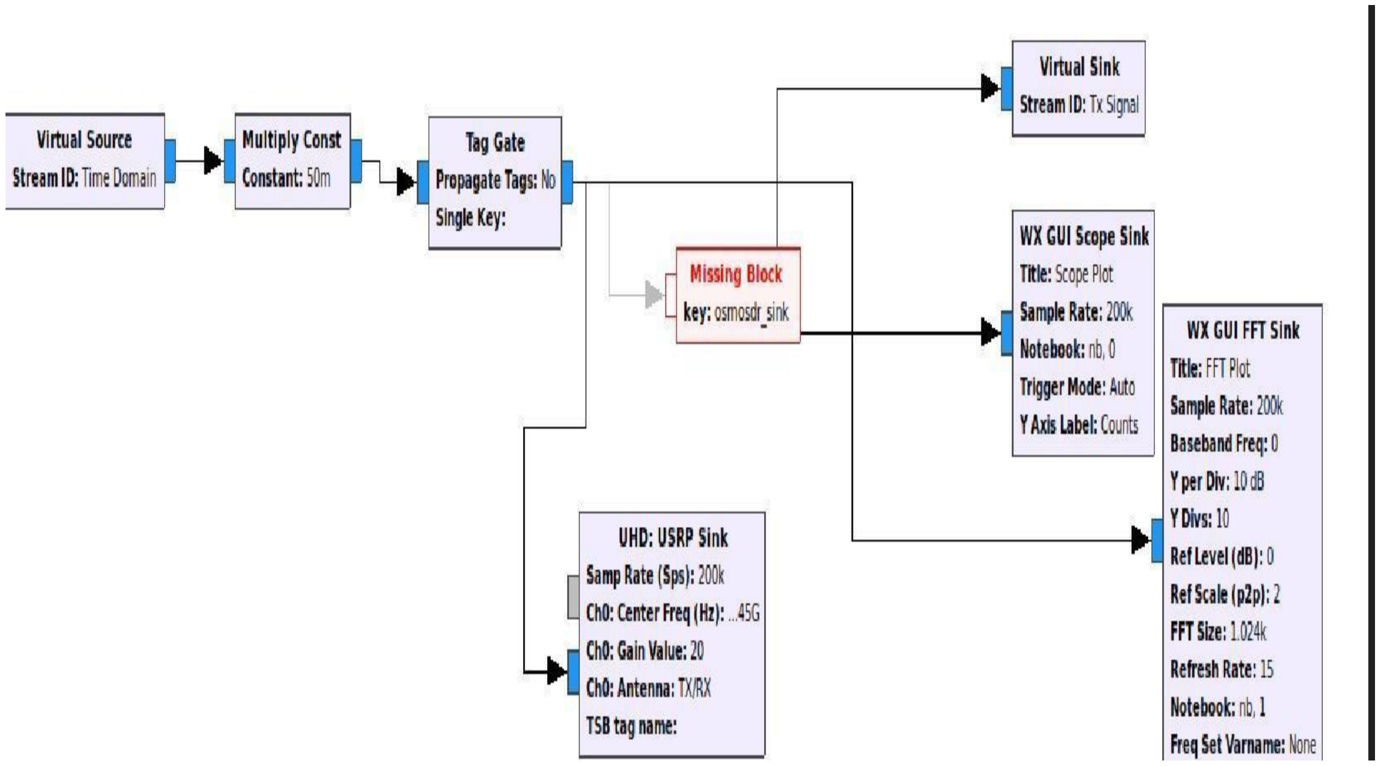
Virtual Source
Stream ID: Header Stream

FFT
FFT Size: 64
Forward/Reverse: Forward
Window:
Shift: Yes
Num. Threads: 1

OFDM Channel Estimation
Synch. symbol 1: sync_word1
Synch. symbol 2: sync_word2
Number of data symbols: 1
Maximum carrier offset: 3
Force One Synchronisation Symbol: No

OFDM Frame Equalizer
FFT length: 64
CP length: 16
Equalizer: <gnura...f44ab0> >
Length Tag Key: frame_len
Propagate Channel State: Yes
Fixed frame length: 1

OFDM Serializer
FFT length: 64
Occupied Carriers: [-..., 26]
Length Tag Key: frame_len
Packet Length Tag Key:
Symbols skipped: 0
Carrier Offset Key:



Virtual Source
Stream ID: Time Domain

Multiply Const
Constant: 50m

Tag Gate
Propagate Tags: No
Single Key:

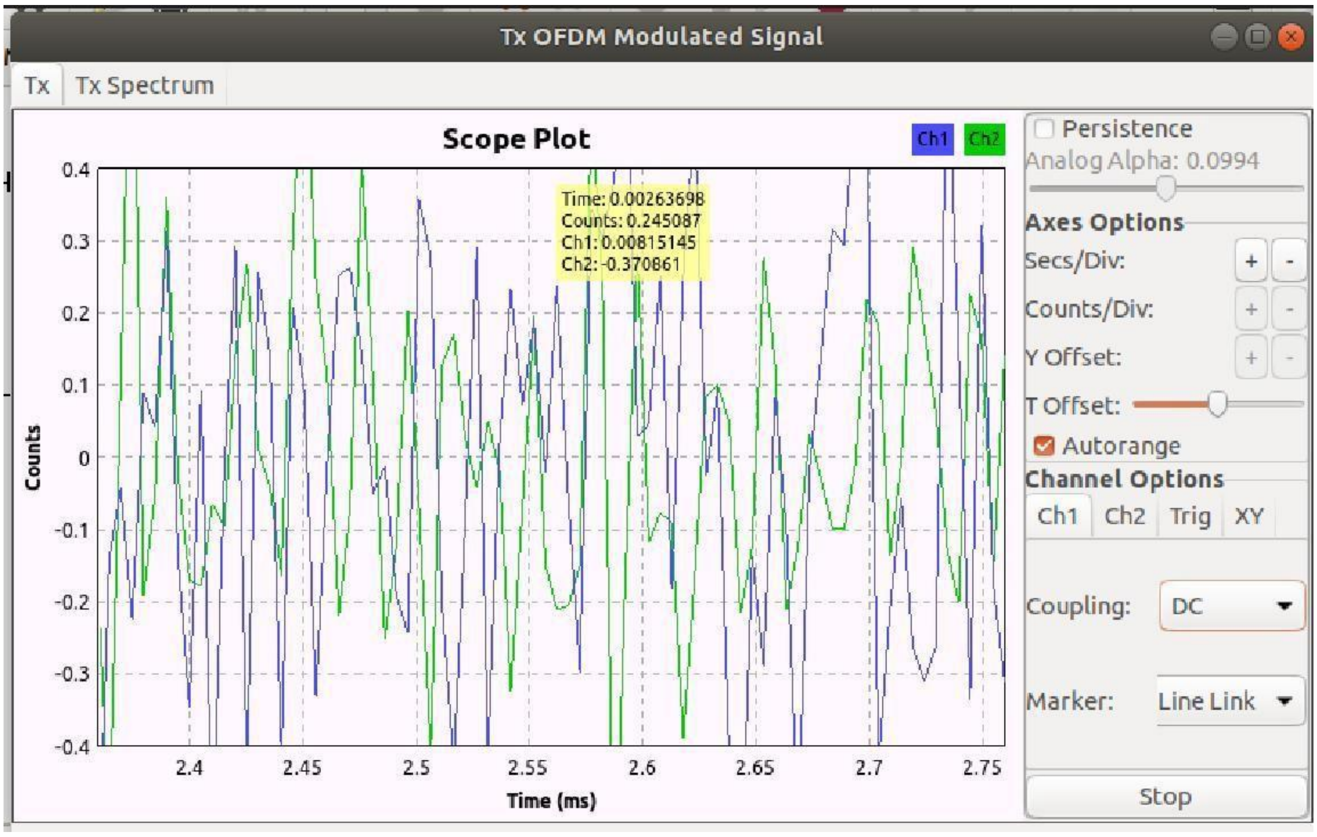
Missing Block
key: osmosdr_sink

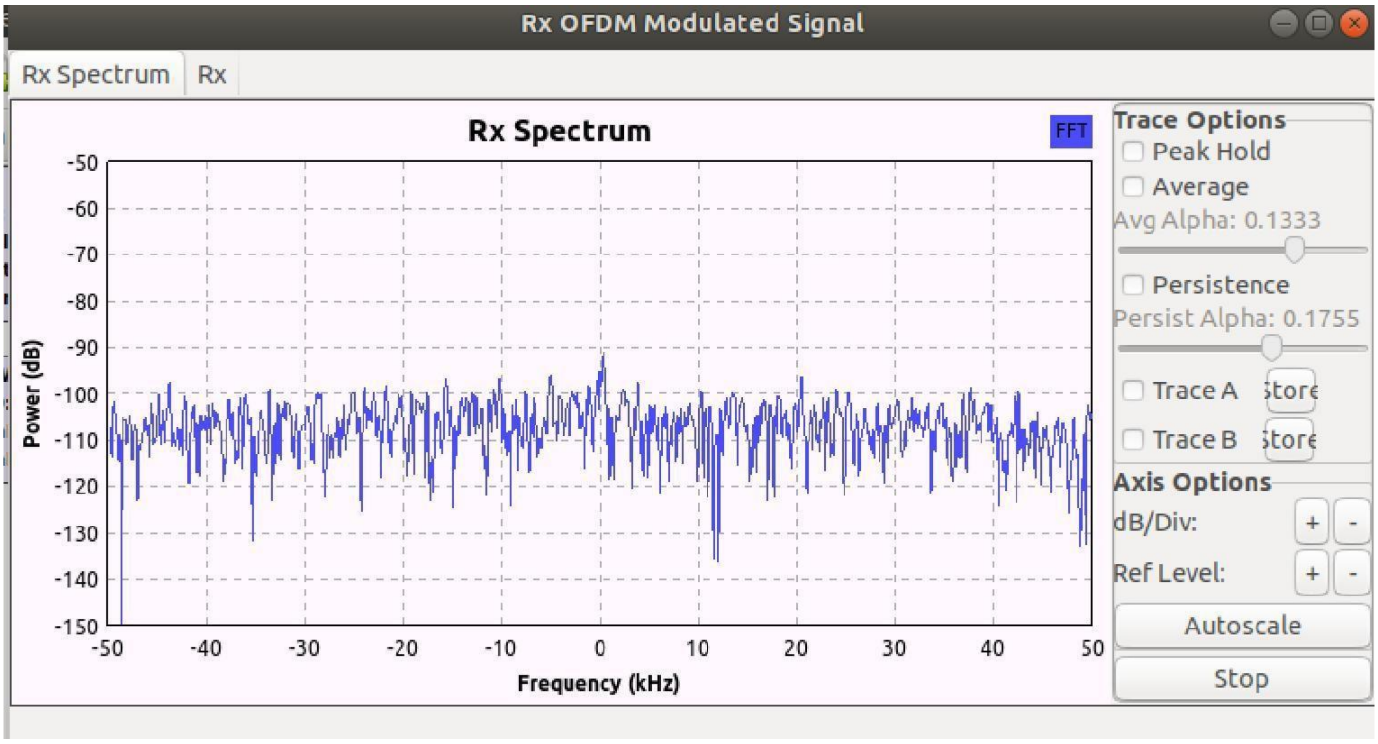
Virtual Sink
Stream ID: Tx Signal

WX GUI Scope Sink
Title: Scope Plot
Sample Rate: 200k
Notebook: nb, 0
Trigger Mode: Auto
Y Axis Label: Counts

WX GUI FFT Sink
Title: FFT Plot
Sample Rate: 200k
Baseband Freq: 0
Y per Div: 10 dB
Y Divs: 10
Ref Level (dB): 0
Ref Scale (p2p): 2
FFT Size: 1.024k
Refresh Rate: 15
Notebook: nb, 1
Freq Set Varname: None

UHD: USRP Sink
Samp Rate (Sps): 200k
Ch0: Center Freq (Hz): ...45G
Ch0: Gain Value: 20
Ch0: Antenna: TX/RX
TSB tag name:


Tx OFDM Modulated Signal
Tx
Tx Spectrum
Scope Plot
Ch1
Ch2
Counts
Time (ms)
Time: 0.00263698
Counts: 0.245087
Ch1: 0.00815145
Ch2: -0.370861
Persistence
Analog Alpha: 0.0994
Axes Options
Secs/Div:
Counts/Div:
Y Offset:
T Offset:
Autorange
Channel Options
Ch1
Ch2
Trig
XY
Coupling:
DC
Marker:
Line Link
Stop


Rx OFDM Modulated Signal
Rx Spectrum
Rx
Rx Spectrum
FFT
Power (dB)
Frequency (kHz)
Trace Options
Peak Hold
Average
Avg Alpha: 0.1333
Persistence
Persist Alpha: 0.1755
Trace A
Store
Trace B
Store
Axis Options
dB/Div:
Ref Level:
Autoscale
Stop

INFERENCE:

A software defined window offers flexibility to deliver the highly reconfigurable system requirements. This approach allows a different type of communication system requirements such as standard, protocol or signal processing method, to be deployed by using the same set of hardware and software such as USRP and GNU radio respectively. However, the realization of SDR concept is inherently limited by analog components of hardware being used.

RESULT and CONCLUSION:

Thus, we have studies the SDR based transceiver system using USRP kit and GNU radio

EXP NO: 11 **DATE:03/04/24**	**Performance Analysis Of QPSK Using SYSTEMVUE**

AIM:

- To study and analyse the QPSK model using SystemVue.

SOFTWARE REQUIRED:

- SystemVue

THEORY:

QPSK:

Quadrature Phase Shift Keying (QPSK) is a form of Phase Shift Keying in which two bits are modulated at once, selecting one of four possible carrier phase shifts (0, 90, 180, or 270 degrees).

The QPSK signal within a symbol duration T is defined as

$$s(t)=A\cdot\cos[2\pi ft+\theta_n],\ 0\le t\le T,\ n=1,2,3,4 \quad \text{-----}\ (1)$$

where the signal phase is given by

$$\theta_n=(2n-1)\pi/4$$

Therefore, the four possible initial signal phases are $\pi/4, 3\pi/4, 5\pi/4, 7\pi/4$.

Equation (1) can be re-written as

$$s(t)=A\cdot\cos\theta_n\cdot\cos(2\pi ft) - A\cdot\sin\theta_n\cdot\sin(2\pi ft)$$
$$=S_i\phi_i(t)+S_q\phi_q(t)$$

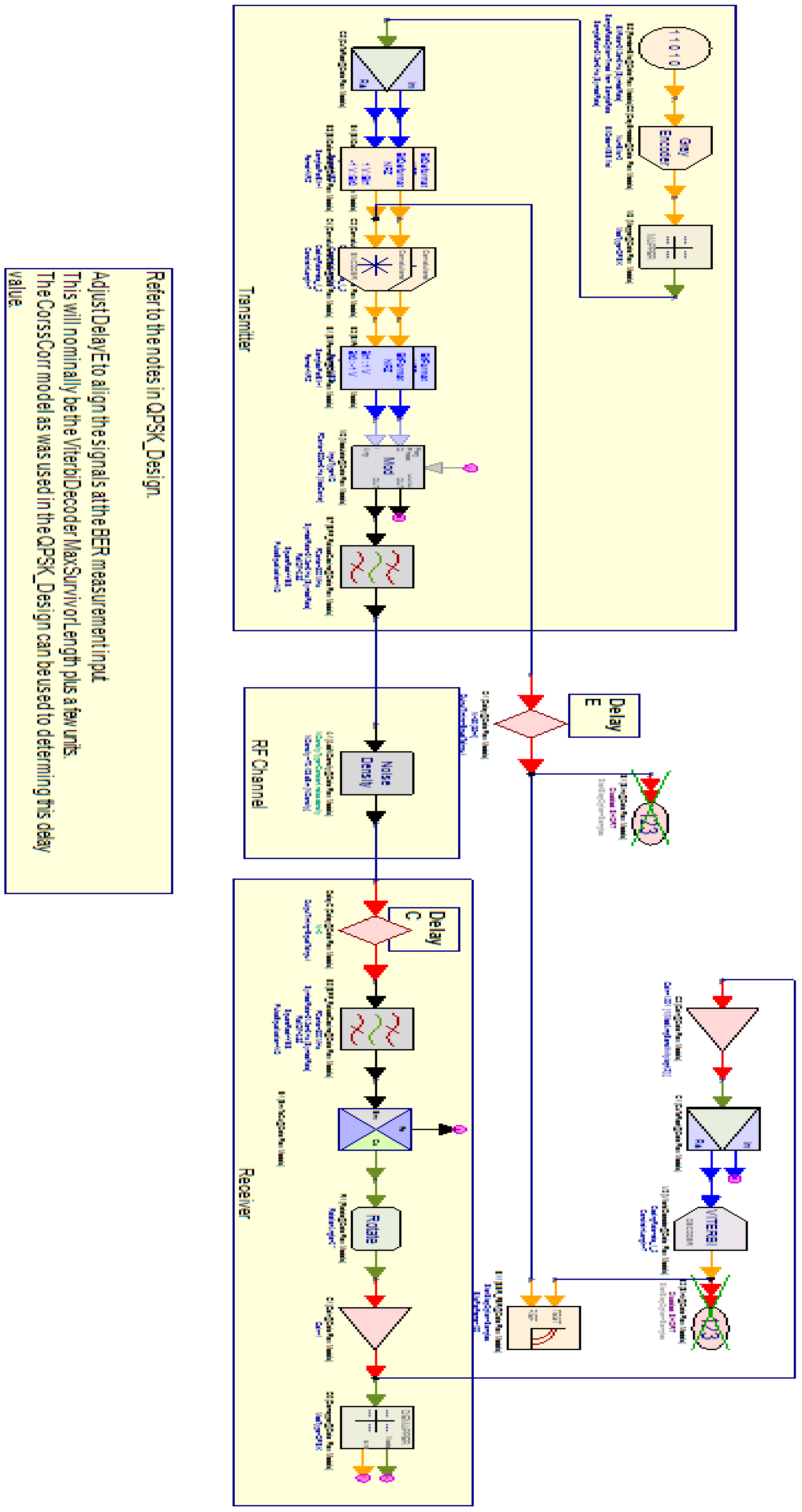
The above expression indicates the use of two orthonormal basis functions: $\langle \phi_i(t), \phi_q(t) \rangle$ together with the inphase and quadrature signaling points: $\langle S_i, S_q\rangle$. Therefore, on a two dimensional co-ordinate system with the axes set to $\phi_i(t)$ and $\phi_q(t)$, the QPSK signal is represented by four constellation points dictated by the vectors $\langle S_i, S_q\rangle$ with n=1,2,3,4.

UNCODED QPSK:

- Uncoded Quadrature Phase Shift Keying (QPSK) is a digital modulation scheme used in communication systems. In QPSK, the information is encoded into the phase of the carrier signal. Unlike Binary Phase Shift Keying (BPSK), which shifts the phase by 180 degrees for each symbol, QPSK shifts the phase by 90 degrees for each symbol.

- In uncoded QPSK, there is no error-correction coding applied to the data before modulation. This means that the raw data is directly modulated onto the carrier signal without any additional redundancy added for error correction purposes.

SYSTEM DESIGN:



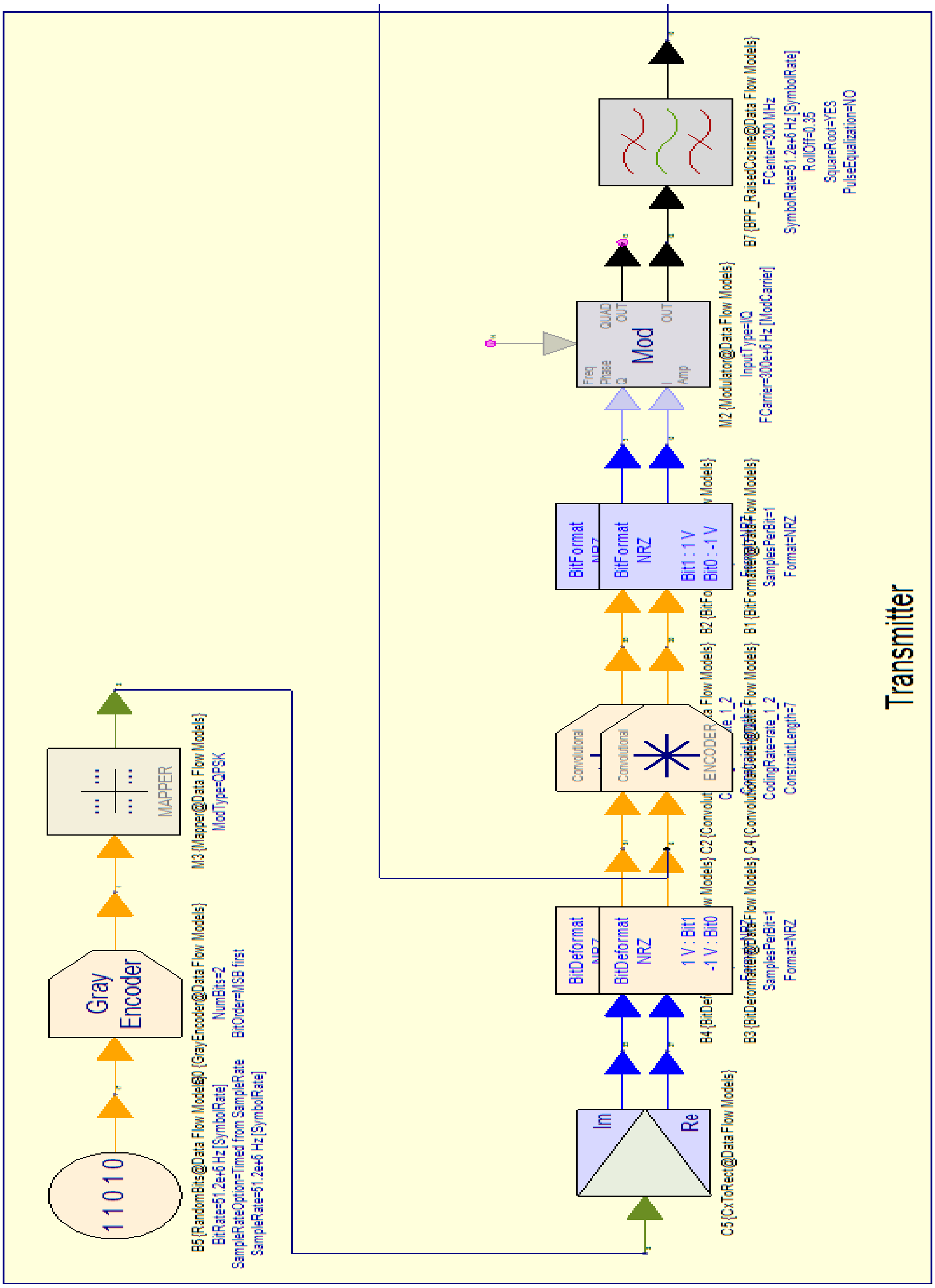
Refer to the notes in QPSK_Design.
Adjust DelayE to align the signals at the BER measurement input.
This will nominally be the ViterbiDecoder MaxSurvivorLength plus a few units.
The CorssCorr model as was used in the QPSK_Design can be used to determining this delay value.

- However, it is more susceptible to errors introduced by noise and interference in the communication channel. As a result, it may be used in applications where error resilience is not a critical requirement, or in conjunction with other error-correcting techniques such as retransmission protocols or channel coding schemes.
- Uncoded QPSK is simpler and more bandwidth-efficient compared to coded schemes because it doesn't involve the overhead of error-correction coding.

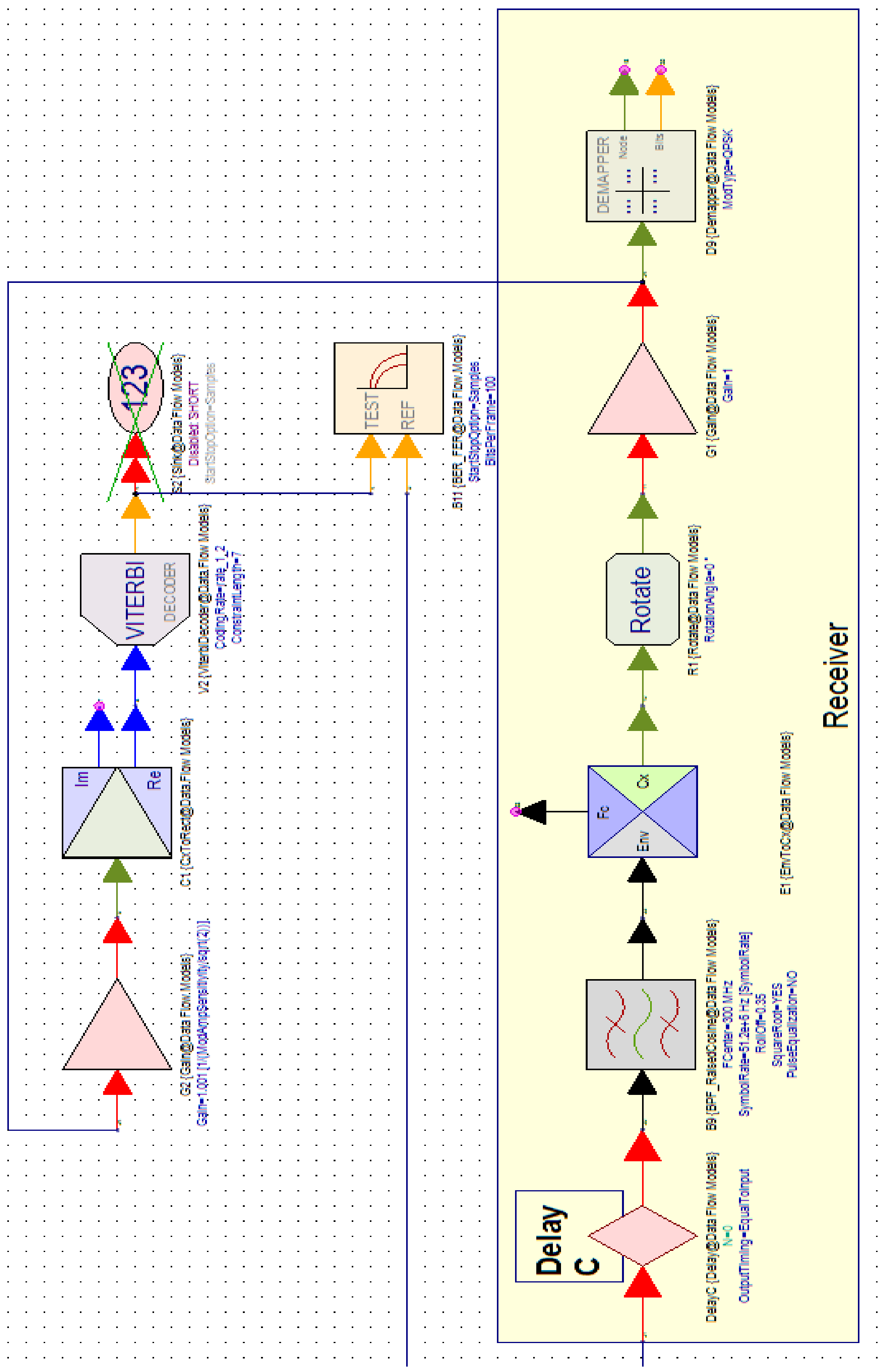
ALGORITHM:

- Open Systen Vue and create new project
- Design the system bu choosing components for transmitter, channel and receiver.
- Simulate and run the project.
- Plot BER vs E_b vs N_o graph by choosing sweep ty[e, start and end points.

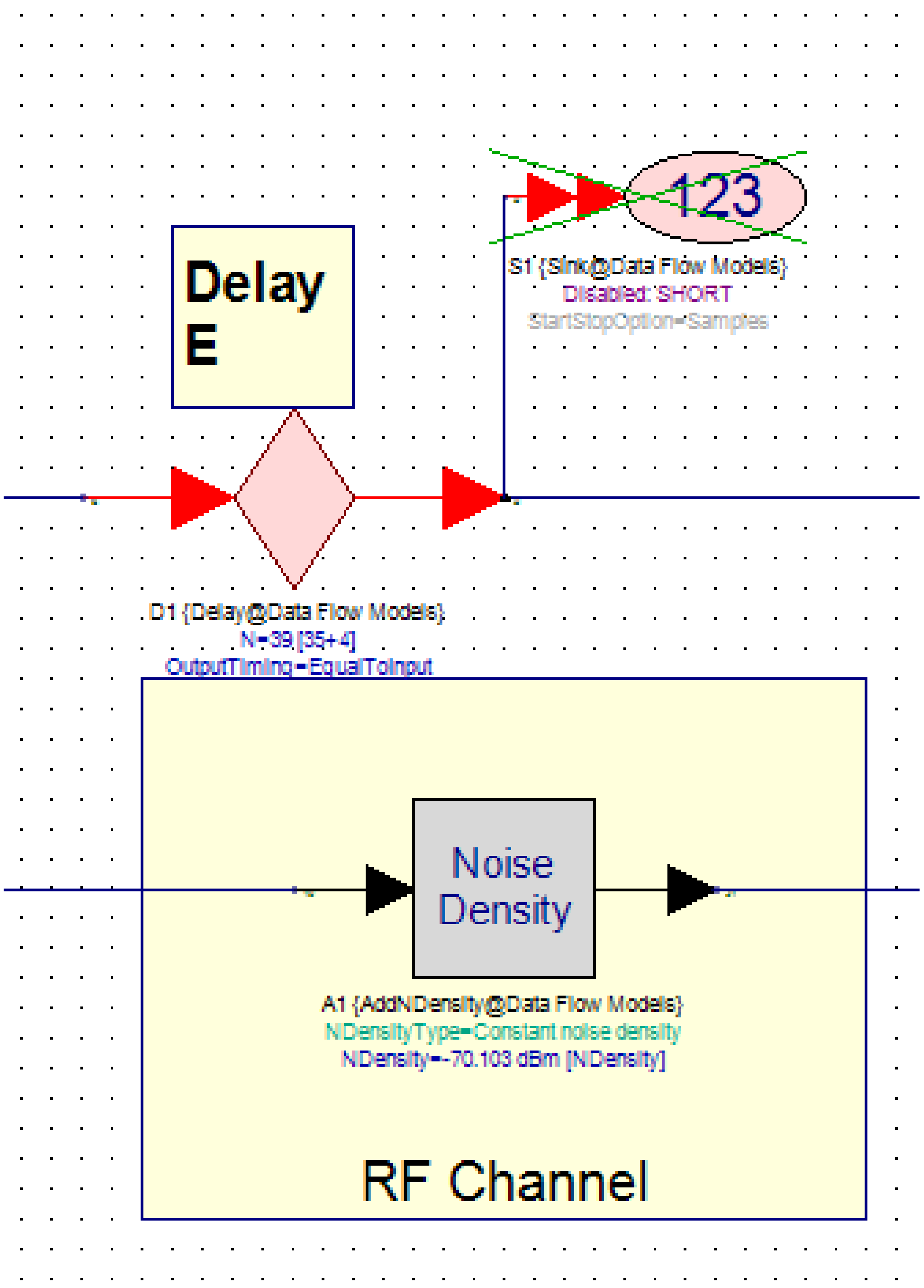
TRANSMITTER:


Transmitter
1 1 0 1 0
Gray Encoder
MAPPER
B5 {RandomBits@Data Flow Models}
BitRate=51.2e+6 Hz [SymbolRate]
SampleRateOption=Timed from SampleRate
SampleRate=51.2e+6 Hz [SymbolRate]
G0 {GrayEncoder@Data Flow Models}
NumBits=2
BitOrder=MSB first
M3 {Mapper@Data Flow Models}
ModType=QPSK
Im
Re
C5 {CxToRect@Data Flow Models}
BitDeformat NRZ
1 V : Bit1
-1 V : Bit0
B3 {BitDeformat@Data Flow Models}
SamplesPerBit=1
Format=NRZ
Convolutional ENCODER
C4 {Convolutional...@Data Flow Models}
CodingRate=rate_1_2
ConstraintLength=7
BitFormat NRZ
Bit1 : 1 V
Bit0 : -1 V
B1 {BitFormat@Data Flow Models}
SamplesPerBit=1
Format=NRZ
Freq
Phase
Q
I
Amp
QUAD OUT
OUT
Mod
M2 {Modulator@Data Flow Models}
InputType=IQ
FCarrier=300e+6 Hz [ModCarrier]
B7 {BPF_RaisedCosine@Data Flow Models}
FCenter=300 MHz
SymbolRate=51.2e+6 Hz [SymbolRate]
RollOff=0.35
SquareRoot=YES
PulseEqualization=NO

RECEIVER:

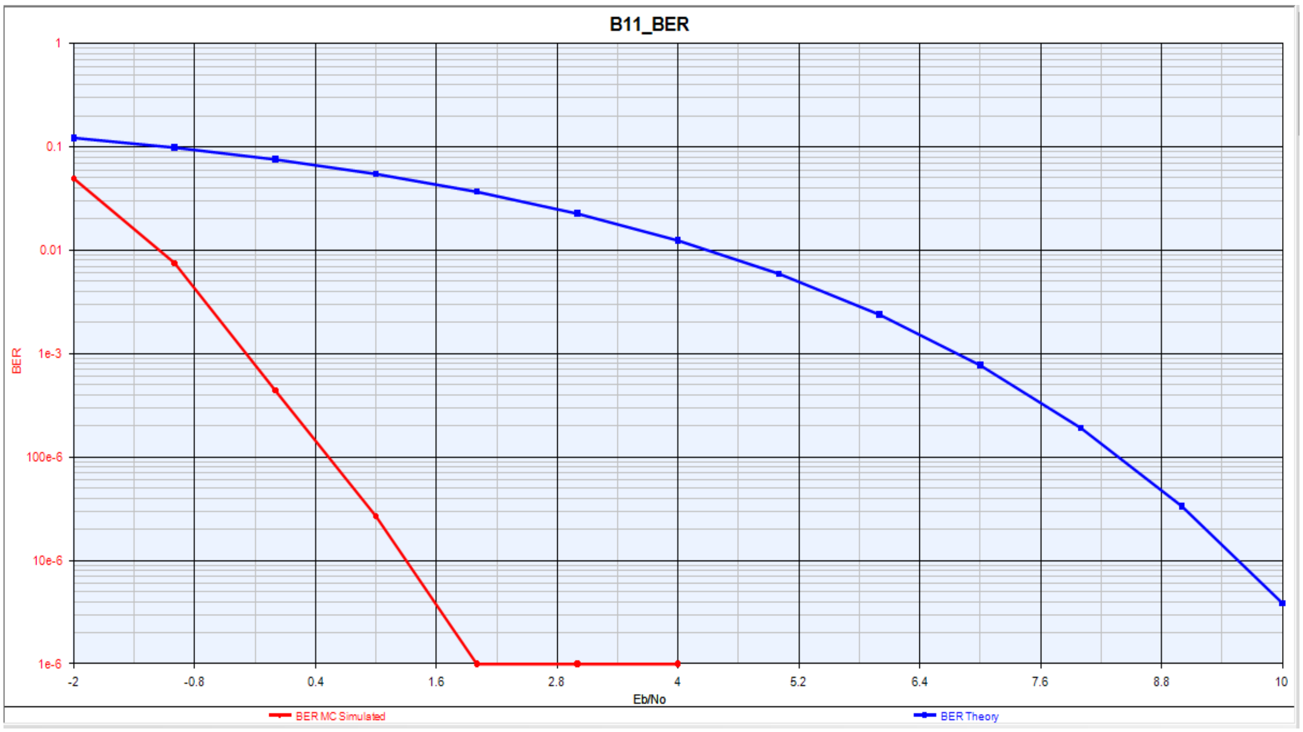

Receiver
Delay C
DelayC {Delay@Data Flow Models}
N=0
OutputTiming=EqualToInput
B9 {BPF_RaisedCosine@Data Flow Models}
FCenter=300 MHz
SymbolRate=51.2e+6 Hz [SymbolRate]
RollOff=0.35
SquareRoot=YES
PulseEqualization=NO
Env
Fc
Cx
E1 {EnvToCx@Data Flow Models}
Rotate
R1 {Rotate@Data Flow Models}
RotationAngle=0 °
G1 {Gain@Data Flow Models}
Gain=1
DEMAPPER
Node
Bits
D9 {Demapper@Data Flow Models}
ModType=QPSK
TEST
REF
B11 {BER_FER@Data Flow Models}
StartStopOption=Samples
BitsPerFrame=100
123
S2 {Sink@Data Flow Models}
Disabled: SHORT
StartStopOption=Samples
VITERBI
DECODER
V2 {ViterbiDecoder@Data Flow Models}
CodingRate=rate_1_2
ConstraintLength=7
Im
Re
C1 {CxToRect@Data Flow Models}
G2 {Gain@Data Flow Models}
Gain=1.001 [1/(ModAmpSensitivity/sqrt(2))]

CHANNELS:

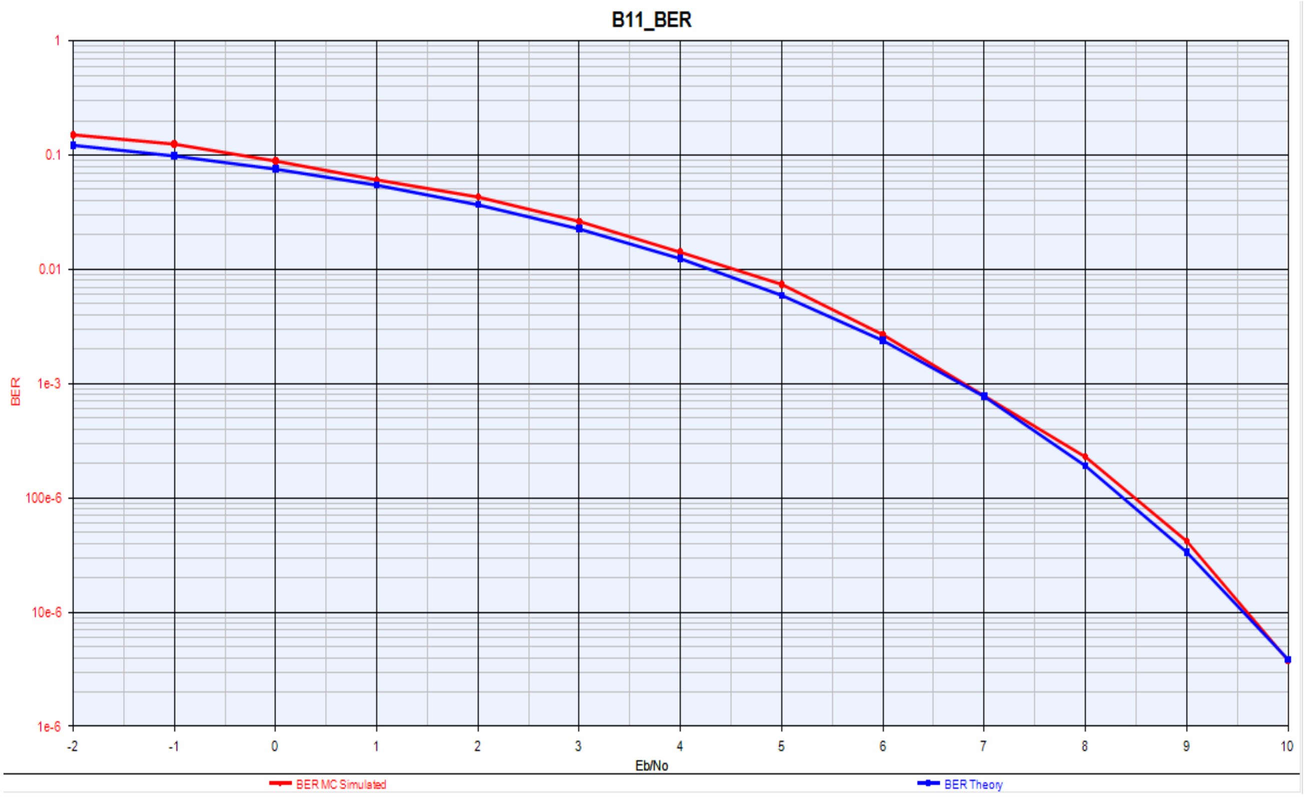

123
S1 {Sink@Data Flow Models}
Disabled: SHORT
StartStopOption=Samples
Delay E
D1 {Delay@Data Flow Models}
N=39 [35+4]
OutputTiming=EqualToInput
Noise Density
A1 {AddNDensity@Data Flow Models}
NDensityType=Constant noise density
NDensity=-70.103 dBm [NDensity]
RF Channel

OUTPUT PLOT:

UNCODED QPSK - BER vs E_b/No



CODED QPSK - BER vs E_b/No



INFERENCE:

- The Graph Coded_BER shows the simulated BER curve with the theoretical BER curve overlaid. The theoretical BER curve is defined in the Equation2 page. This example demonstrates the substantial BER performance improvement when convolutional coding is used and soft decision detection is used.
- The Graph Uncoded_BER shows the simulated BER curve with the theoretical BER curve overlaid. The theoretical BER curve is defined in the Equation2 page. Excellent agreement is obtained.

RESULT:

Thus, QPSK performance has been studied and analysed using SystemVue.